
Rockford Fosgate
RTTI

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Introduction

Preface: An Introduction to Rockford Corporation

Rockford Corporation is an audio electronics company which manufactures and sells audio amplifiers, signal processors and speakers. Rockford products are distributed under two brand names. The Rockford Fosgate products are the most popular in car audio, prized for their rugged high performance and stunning good looks. The Rockford Fosgate speakers are designed and manufactured in Michigan by Rockford Acoustic Designs (RAD). While operating under their former name, Carbonneau speakers, RAD was known as a major Original Equipment Manufacturer who supplied speaker components to some of the most respected names in professional and home audio. Under the Hafler brand name Rockford supplies professional equipment to recording studios, sound contractors and movie theaters. Hafler amplifiers deliver a level of sound quality and reliability which has made them one of the most popular monitor amps in the very demanding studio market. How is it possible for one company to command the users' respect in such widely divergent applications? In a word – *innovation.*

Rockford engineers have never been content to accept conventional wisdom and have always questioned the underlying assumptions of circuit implementation. The application of original research of the acoustic properties of auto interiors led to the development and subsequent patenting of the Punch Equalization circuit. The Punch Equalization revolutionized the infant car audio industry by demonstrating that high quality sound reproduction could be achieved inside a car. Over the years many of Rockford's innovations have been adopted as industry standards. Rockford engineers pioneered and refined the use of power MOSFETs in car, home and professional amplifiers.

Over the course of the last few years, this need to improve and refine the products has been applied to our electronics manufacturing processes. The implementation of efficient "Just in Time" purchasing procedures, and the use of automated circuit assembly and surface mount components ensures continued improvements in efficiency and cost control. This guarantees that Rockford products will maintain their characteristically high value into the future.

Rockford Services

Rockford Corporation 800-366-2349 Fax 602-966-3983

The Rockford Corporation offers professional dealer support to assist you with anything you may require. Rockford has three separate 800 telephone lines which provide immediate service from 8:00 a.m. to 5:00 p.m. Mountain Standard Time, Monday through Friday.

Sales Department 800-366-2349 Fax 800-366-6724

The Sales Department can assist you with any questions you may have with your account or any product information such as current pricing or employee purchase.

Dealer Technical Assistance 800-743-3526 Fax 602-967-2807

This line is available only to Authorized Rockford Fosgate Dealers and is maintained by Rockford's Technical Support. A voice mail service is available to leave a message during non-business hours.

Customer Service / Parts Dept 800-669-9899 Fax 800-398-3985

This department can assist you in ordering parts and warranty repair on any Rockford Fosgate product.

Purpose for R.T.T.I. Training Program

The primary purpose for this training program is to help you sell more Rockford Fosgate products. You'll learn the differences between our product and other brands, we know you will choose to sell Rockford Fosgate. You will also learn more about audio and become better skilled as a salesperson, increasing your income and the success of your business.

Many of the concepts we discuss are very basic. Since everyone in the audio business is at a different level of knowledge, the information presented is from its simplest form to the more advanced. This program is not designed to make engineers of salespeople. It is, however, designed to help you "Think on Your Feet." You'll be able to deal with objections and questions as if you've rehearsed each presentation prior to the customer walking in the door.

In order for you to get to this point, ***you must read this manual.*** Your instructor will highlight the most important points, but reading the entire manual will help you digest the information. Learn as much information as you can so that it can be used for your own benefit.

Jim Fosgate Founder

Rockford Fosgate's beginning started with Jim Fosgate's dream of high performance car audio. When the first Fosgate PR-7000 automotive amplifier was shown at the Chicago C.E.S. in 1973, there weren't any head units on the market with line level outputs and there weren't any dealers willing to order a $300 car amp. This didn't stop Jim from pursuing his goals. He had found a better way and he was determined to educate the market.

Jim Fosgate's passion for audio was nurtured by a friend and avid hobbyist, a dealer who lived in the mid-west. While Fosgate's company Pro-Line in Salt Lake City was building radio transmitters and receivers for remote control airplanes, Jim Fosgate was developing a circuit called a "Frequency Energizer."

The "Frequency Energizer" circuit was developed for Fred Hulan's *Audio Mart* in Kansas City. Fred was doing professional installations in homes and theaters using a then state-of-the-art piece of test equipment called an "Acoust-A-Voice." It incorporated pink noise, a 1/3 octave equalizer and an oscilloscope. Fred kept notes on the inefficiencies he saw repeatedly in each installation. He shared those findings to Jim Fosgate whose Frequency Energizer circuit would compensate for loss in the high and low end frequencies. (Starting to sound familiar?) This first unit was designed to run through a tape loop in the receiver or preamplifier.

The birth of Fosgate car audio came when the Frequency Energizer was incorporated into a 30 watt car amp by Jim and some of his after hours enthusiasts. These late night experiments lead to the PR-7000.

Rockford Corporation

By the end of the seventies, Jim Fosgate had developed interests in other businesses and was pretty much letting the company run and manage itself. Financial troubles gave way to quality problems and Mr. Fosgate began looking for a buyer for his innovative car audio company.

Rockford Corporation was formed as an investment group. Their organization at the time was called Camelback Investment Group. They adopted the name of the street where Fosgate audio had existed for years in Tempe, Arizona. Their goal was to take a floundering company, turn it around, and make it an industry leader.

The biggest initial change made at the newly formed Rockford Fosgate was the strict reins put on quality control. Key people were put into positions and responsibilities that assured total quality control procedures. Many of the key individuals in these positions are still top executives at Rockford Corporation today.

One of these key people was an engineer named John France. John had actually been working with Jim Fosgate since the mid seventies. It wasn't until Rockford's involvement that John France's position became so important. It was John's interpretation of Jim Fosgate's designs that made Rockford Fosgate's early products so successful. Innovative ideas such as bridging mono while running stereo and the use of MOSFET's for better stability at low impedance loads are proof of John's vision. John also developed the protection circuit that is now used in all Punch and Series One amplifiers. This circuit was first developed for the Power 1000 and its effectiveness was demonstrated to R.T.T.I. students by burning pencils and welding with foil without damage to the amplifier.

As Rockford Fosgate products became more in demand, a peculiar problem developed. Dealers would call the factory and say, "We love your amps, but they're blowing up every speaker we have." So the next hurdle to overcome was to develop a high performance speaker line.

After evaluating many samples from outside speaker manufacturers, it became evident that nobody could deliver what Rockford Fosgate wanted. At that point it was determined that the only way to make a speaker worthy of the Rockford Fosgate name was to build it ourselves. Rockford found a quality O.E.M. speaker manufacturing in Grand Rapids, Michigan, which had been in the audio business since the early 1940's. Carbonneau speaker manufacturing had focused their business on the O.E.M. of mobile speakers. Carbonneau was excited about being involved with Rockford Corporation in the production of hi-performance car audio. In 1985, Rockford Fosgate began building its new line of car audio woofers, mid ranges, and tweeters. Rockford Carbonneau made a quick transformation from O.E.M. to after-market.

Rockford Technical Training Institute

In 1986 Rockford Corporation started a program to teach installers the best way to use Rockford Fosgate amplifiers and speakers. The course started with system design, passive crossover design and the basics of Speaker enclosures. A second, more advanced class was also developed and Rockford's Technical Training Institute was now in full scale operation. Still to this day there are those who can't believe that one amp can run twenty speakers. Rockford Technical Training Institute has helped Rockford dealers stay several steps ahead of the competition.

Car Audio Contest Trilogy

Rockford Corporation has always believed that the more our industry prospers, the more Rockford will prosper. For this reason Rockford Corporation has always been a supporter of I.A.S.C.A. Rockford employees co-authored the first National Autosound Challenge Association (N.A.C.A.) judging sheet and rule book. This organization eventually became the International Autosound Challenge Association.

In 1985, Rockford Fosgate marketing wanted to take the concept of contests that was being promoted by Doc Thunder and put it into a format that would expand the awareness of quality high-end car audio.

With the help of John France, Rockford Fosgate created, outlined and defined the goals of sound quality, installation and weighted sound pressure level as determining factors of a winning system. This original sheet was very simple and easy to use. I.A.S.C.A. uses very much the same scoring sheet to this day.

Rockford Fosgate created the challenge package to help dealers learn how to promote contests. At this time Rockford Fosgate even supplied the judging sheets and the judges. Time went by and everyone saw the value of promoting through contests. Rockford Fosgate decided to share our experience in this sort of promoting to the entire industry in a meeting at the WCES. We felt it would benefit the entire country if all manufacturers supported the same judging format.

This led to the formation of N.A.C.A. and later I.A.S.C.A. We dreamed of our efforts turning into something like N.H.R.A., but to see it actually happen has been like watching history being written.

Team America Program (I.A.S.C.A / U.S.A.C.)

Rockford Fosgate announces an updated Team America Program designed to significantly improve support of competitors using Rockford Fosgate products. This program is open to anyone who purchases Rockford Fosgate equipment.

Team America members must be active members of IASCA or USAC. If the individual is not a member of one of these organizations, Rockford will provide applications to join. Team America members will receive a Team America t-shirt, product patch, Wearables brochure, car sticker, membership card, and the Team America Magazine, "Made in the U.S.A." Members can earn points towards several unique Team America products through competition in IASCA or USAC competition. If a Team America member qualifies and competes in either an IASCA or USAC National Final, that member will receive special prizes.

THE ROCKFORD EPIC

1969-1973	Jim Fosgate's company *Pro-Line* builds radio transmitters and receivers used in remote control airplane models. Audio is just a hobby.
1973	Jim Fosgate designs home EQ (Frequency Energizer) unit to compensate for inefficiencies in home and theater listening environment.
1973	Jim Fosgate's company, Pro-Line, puts Frequency Energizer into 30 watt per channel car amp. The patented Punch EQ circuit is born.
1973	Summer Consumer Electronics Show. The newly formed auto sound company, Fosgate Audio, showed a revolutionary amplifier with unique Punch EQ. First model was the PR 7000.
1975	Fosgate begins manufacturing amplifiers with discrete components rather than integrated circuits. John France is involved with engineering of new products.
1980	ROCKFORD CORPORATION is formed and invests in Jim Fosgate's company and ideas.
1980	Jim Fosgate leaves *Fosgate* to pursue other ventures including his radically innovative audio/video surround sound processing.
1982	Product quality & reputation improves dramatically.
1983	Thunder On Wheels holds first Rockford Fosgate *"Crank 'em Up"* in Texas.
1985	Rockford Fosgate introduces Punch separates and woofers.
1986	Rockford Technical Training Institute (R.T.T.I.) is formed.
1986	Rockford Corporation acquires Carbonneau Speakers, primarily an O.E.M. builder of speakers.
1986	Pro Series introduced at W.C.E.S.
1990	Rockford Fosgate introduces the Box That Rocks at W.C.E.S.
1991	Symmetry is unveiled at Winter Consumer Electronics Show.
1993	New Punch DSM amplifiers and RFA Audiophile midrange and tweeters unveiled at WCES.
1994	New Punch "i" and "ix" series amplifiers; SYMMETRY EPX; 3X / 5X Signal Processors; Power Series Components; Punch Classic Twins; Punch Classic Woofers and The New Box That Rocks introduced at WCES.
1995	New Power 250m^2, Punch 4020DSM, EPX^2 with 1/3 octave equalization capability. A BLT (Balanced Line Transmitter) will interface with 250m^2 and EPX^2, New "Box That Rocks" with Tubes, New Punch Splits Component Speakers, New Rockford Fosgate Punch Woofers and Series 1 Woofers.

Chapter 1
System Design the Rockford Way

- Crossover Points
- Using "Q"
- Equalization
- Capacitors & Coils
- Crossovers
- Crossover Slopes
- Ohm's Law

System Design The Rockford Way

(How Rockford Dealers can rise to the top)

Satisfy Your Customer's Needs

To best satisfy the customer's expectations of his/her sound system, **LISTEN TO THE CUSTOMER**. Only by qualifying the customer's needs, can a dealer design a system that meets or exceeds the customer's expectations. Give the customer what they need – not what they ask for.

Going the Extra Mile

To separate Rockford dealers from the rest, **maximize the performance** of all systems regardless of system complexity to ensure reliability and sonic performance.

Building for the Future

To benefit the customer and to promote future sales, design systems with expandability in mind. Customer's money spent on equipment that cannot be used in future systems is money wasted.

Building A System

Listed below are some items to consider when designing a system for a customer.

☛ **SPEAKERS**

1. Size
2. Performance
3. Location and type of vehicle

☛ **AMPLIFIERS**

1. Compatibility for future system design
2. How much power is needed for the speakers
3. Current available from the battery

☛ **CROSSOVERS**

1. Passive or Electronic
2. 2-Way, 3-Way, or 4-Way System Design
3. Crossover frequencies
4. Subsonic filters

☛ **EQUALIZATION**

1. Passive or Electronic
2. 1/2 Octave or 1/3 Octave Electronic Equalization
3. Passive Networks

How Speakers Operate

To accomplish stereo imaging inside a vehicle some understanding of speakers and their placement is needed. A speaker has a voice coil and a fixed magnet which interact with each other producing motion from the reaction of AC voltage. The voice coil is attached to the cone and its motion changes pressure in the listening environment. This change in pressure makes a wave pattern that we distinguish as sound. These waves are periodic and complex because music is dynamic.

How Speakers Interact within the Vehicle

Vehicles are plagued with many reflective surfaces and proper speaker placement may be difficult in some vehicles. Installing speakers in a car is fairly easy but having them image well is something much more complex. Mounting locations of component speakers are most critical because of the speaker's operating frequencies and the wavelength of the frequencies they generate make them directional.

Path Lengths

A vertical array of speaker placement is highly effective in the automotive environment. Their path length would ideally be equal distant and on the same x, y axis to the listener. (Figure 1.1)

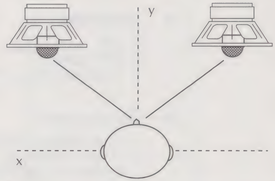


Figure 1.1

Speaker Placement

Speakers require a baffle from the front of the speaker to the rear to eliminate cancellation. The speakers should be then tested in the car to find proper placement and angled to achieve a stable stereo image. Boundary reflections (door panels, people, consoles) can lower the response of the speakers, so going to the effort in verifying placement will yield better results.

AMPLIFIERS

An amplifier is a device that increases the amplitude of a signal. An ideal power amplifier is one who's output signal is identical to its input signal except for the increase in amplitude. The amplifier converts the DC energy from the vehicle's electrical system to AC energy. DC energy is needed to operate a speaker. A stable supply of power is needed from the vehicle's electrical system for an amplifier to operate to its maximum potential. Power is the product of the combination of voltage and current. To increase the power output of an amplifier, the load impedance can be decreased to the minimum load recommendation of that amplifier. In a perfect world, halving the impedance would double the power output of the amplifier. This is why when bridging an amplifier, the power rating increases. You are now using both channels to drive a common load. Therefore, each channel of the amplifier will see half the impedance of the load. Now the power is the sum of left plus right multiplied by two. Verifying the power rating, impedance and quantity of speakers used in the system will dictate the power output the system will produce. For higher output, minimize the load impedance, within recommended limits. If the load impedance is already at minimum, a more powerful amplifier or multiple amplifiers are required.

CHOOSING CROSSOVER POINTS

After finding the best mounting location for the speakers, determine the proper crossover point and Q. The crossover points are determined by the resonance of the driver and also their response in the vehicle. When using a crossover in high-pass on the midrange and tweeter, a one or two octave range from resonance is suggested. For example, an ND4 tweeter has a resonance of 1,500Hz; therefore, a 4kHz crossover point at 12dB/octave will avoid phase disturbance.

USING "Q" TO YOUR ADVANTAGE

The Q of the crossover point is relative on the mounting configuration of the midrange and tweeter. A second order (12dB/octave) Butterworth crossover point has a Q of .707 which is designed for midrange and tweeter to be mounted on the same plane and in a close proximity from one another and on axis to the listening position. This crossover alignment is less sensitive to driver offset and when combined with high and low-pass they will sum flat from their –3dB crossover point. A Linkwitz-Riley has a Q of .49 and therefore is less sensitive to driver offset from a tweeter to a midrange. The crossover point is –6dB at the low and high-pass intersection and will sum flat. This will enable the midrange and tweeter to be mounted on the same plane but further apart from one another. A second order Bessel crossover point has a Q of .58 and is designed for midrange and tweeter crossover points that are altered by a factor of 1.1. An example of different crossover points; 3kHz low-pass 4.110kHz high-pass, these would be utilized to smooth the response of the speakers mounted more than 12" from one another and not on the same plane. If the crossover is sharing the same low and high-pass crossover points, the response will not sum flat. These different crossover Q factors are designed to compensate for vehicle acoustical anomilies and the speaker placement.

EQUALIZATION

- **PASSIVE EQUALIZER**
- **ACTIVE EQUALIZER**

An equalizer is designed to give boost or cut at a given frequency. The use of equalization in a car audio system is to compensate for the acoustic effects within a vehicle and the speaker's frequency response. An equalizer can be used to assist in sound quality of the system. An equalizer is comprised of multiple filters which are frequency selective networks using resistors, capacitors, inductors, transistors, and integrated circuits. There are two types of equalizers: **Passive** and **Active**.

A **Passive Equalizer** does not require power to operate. Passive equalizers (also known as parallel trap filters) use resistors, capacitors, and inductors but no active components. An **Active Equalizer** requires power to operate and uses active components such as transistors and ICs. There are many other types of active equalizers such as:

Rotary Equalizers which use selectable knobs to select frequency range and the degree of boost or cut.

Parametric Equalizers are like rotary equalizers but with added control of the center frequency and bandwidth of the other frequency bands.

Graphic Equalizers have individual control of boost or cut over the amount of bands on the equalizer. Each band can be spaced from full octave to half octave to one third octave depending on the amount of control that is needed in that bandwidth. A **one octave** band equalizer means that each band will be spaced one octave away from one another. For example, 100Hz is one octave above 40Hz. This equalizer is usually limited in the amount of control because of the small amount of bands used. A **half octave** equalizer gives more control because the bands are a half octave distance. For example: 45Hz is a half octave above 32Hz. This type of equalization may be used in the lower band frequencies of a full octave equalizer to give added bass control. A **one third octave** equalizer has the ability to control more frequency bands than other graphic equalizers because of the amount of filter bands that can be used. The frequency bands are very narrow and have a constant "Q." For example, the frequency band differences are 25Hz to 31.5Hz and 31.5Hz to 40Hz. Generally, more bands have a better ability to control the response. Equalizers are not always the answer in some applications when you may only have need to position the speakers differently or apply more power or even change crossover points. Equalizers can actually function the best when they are used in a cut configuration rather than a boost for good sound quality systems.

Polystyrene Capacitors
- Low tolerance

Mylar Film Capacitors
- Small Values

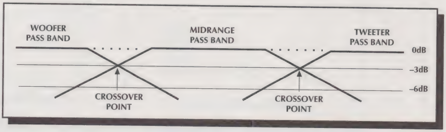
Electrolytic Capacitors
- Low Cost

Capacitors pass high frequency information and block low frequency information. A capacitor is a device which will store an electrical charge. It consists of metal plates which are separated by an insulating material (dielectric). When using non-polarized capacitors the dielectric acts as a blocking device against DC current. When AC current is applied, the capacitor will conduct the current to the amplifier. Capacitors will attenuate frequencies below the cutoff frequency, but pass frequencies at and above the cutoff frequency with little attenuation. There are many different types of capacitors; **polystyrene**, **mylar film**, and **electrolytic**. These capacitors come in different **tolerances**, **values** and **operating voltages**. Generally, in mobile electronics we deal with non-polarized electrolytic capacitors simply because they're easy to obtain and are fairly inexpensive.

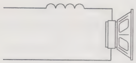
Air Coils
- High Power Handling

Iron Coils
- Small Size

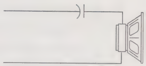
Inductors, sometimes called coils, pass low frequency information and block high frequency information. The function of the coil is to increase its AC (resistance) as frequency increases. A coil is made of windings of wire and has an inductance when carrying AC current. These windings of wire can have an air or iron core which will determine the inductance of the coil. **Air core coils** have no magnetic material in their core and rely on the amount of windings of wire to create their inductance. This will enable them to have high power handling and lower change in creating a magnetic field. **Iron core coils** utilize a magnetic material to work in conjunction with the turns of wire. This can create a smaller coil but a higher change in creating a magnetic field.



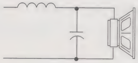
Passive Crossovers



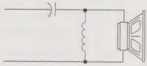
6dB/Octave Low-Pass



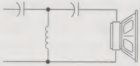
6dB/Octave High-Pass



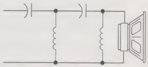
12dB/Octave Low-Pass



12dB/Octave High-Pass

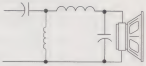


18dB/Octave High-Pass

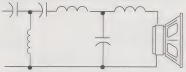


24dB/Octave High-Pass

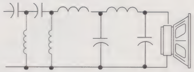
6dB/Octave Bandpass



12dB/Octave Bandpass

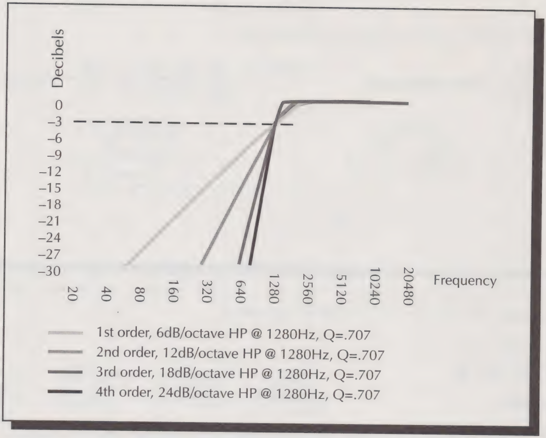


18dB/Octave Bandpass



24dB/Octave Bandpass

Crossover Slopes

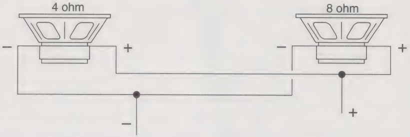


Ohm's Law

Parallel Wiring Formula for 2 Speakers

$$\frac{R1 \times R2}{R1 + R2} = \textbf{LOAD IMPEDANCE}$$

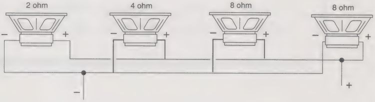
EXAMPLE: $$\frac{\text{4 Ohms times 8 Ohms}}{\text{4 Ohms plus 8 Ohms}} = \frac{32}{12} = \text{2.67 Ohms}$$



Parallel Wiring for Multiple Impedances

$$\frac{1}{\frac{1}{R1} + \frac{1}{R2} + \frac{1}{R3} + \frac{1}{R4}}$$

EXAMPLE:



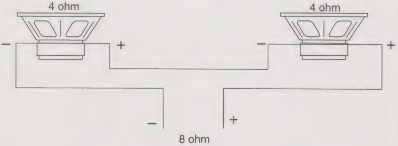
Drivers' Impedance in Fraction form

$$\frac{1}{\frac{1}{2} + \frac{1}{4} + \frac{1}{8} + \frac{1}{8}}$$

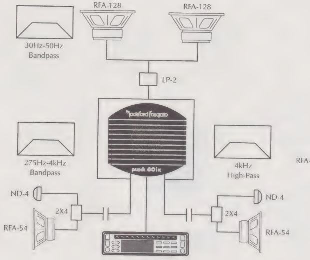
Convert to Decimal form and add together $$\frac{1}{.5 + .25 + .125 + .125} = \text{1 Ohm}$$

Series Wiring Formula

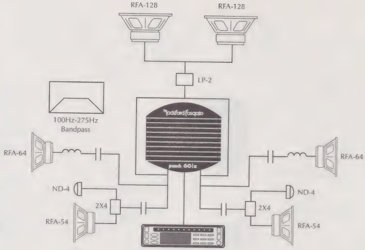
R1 + R2 = LOAD IMPEDANCE



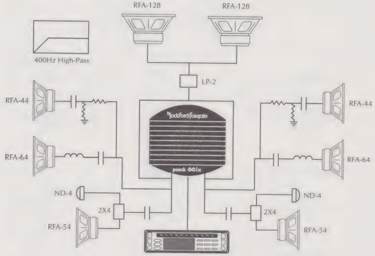
System Design



Crossover Points: (30Hz High-Pass 60ix)
50Hz Low-Pass / 275Hz High-Pass
Products Used: Punch 60ix
RFA-514
LP-2
(2) RFA-128
(2) 145 µF caps
System Configuration: 3-Way System Mixed Mono Mode

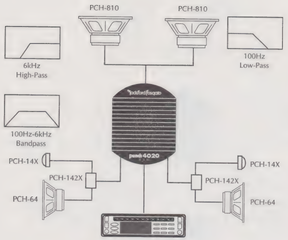


Crossover Points Added: 50Hz - 275Hz Bandpass
Products Added: (2) RFA-64
(2) 253 µF caps
(2) 3.6 mh coils
System Configuration: 4-Way System

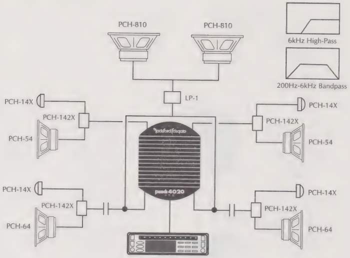


Crossover Points Added: 400Hz High-Pass
Products Added: (2) RFA-44
(2) 100 µF caps
(2) 1Ω 20 Watt resistors
(2) 10Ω 20 Watt resistors
System Configuration: RFA-44 for Rear Fill and –3dB L-Pad
1Ω resistor in series
10Ω resistor in parallel with speaker

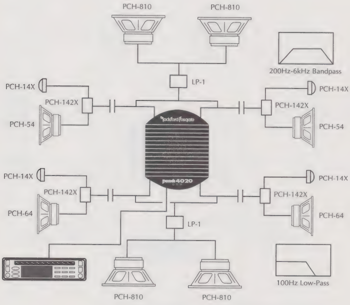
System Design



Crossover Points: (100Hz High-Pass
100Hz Low-Pass 4020)
6kHz High/Low-Pass PCH 142X
Products Used: Punch 4020DSM
PCH-614
(2) PCH-810
System Configuration: 3-Way System – 3 Channel Mode

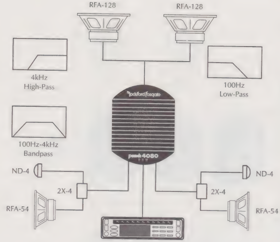


Crossover Points Added: 30Hz High-Pass/200Hz High-Pass
100Hz Low-Pass
Products Added: PCH-514
LP-1
(2) 200 µF caps
System Configuration: 4-Channel Mode
Front/Rear Fade Subwoofers
in Mixed Mono on Front Output

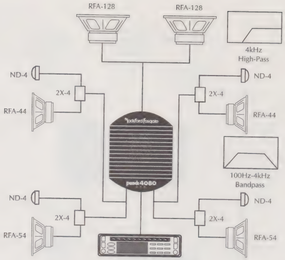


Crossover Points Added: 100Hz Low-Pass / 200Hz High-Pass
Products Added: LP-1
(2) PCH-810
(2) 200 µF caps
System Configuration: Increased Bass Impact
Additional Pair Subwoofers in
Mixed Mono on Rear Output

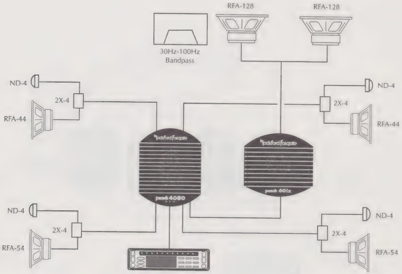
System Design



Crossover Points: (100Hz Low-Pass
100Hz High-Pass 4080)
4kHz High/Low-Pass
Products Used: Punch 4080DSM
RFA-514
RFA-128
System Configuration: 3-Way System – 3 Channel Mode

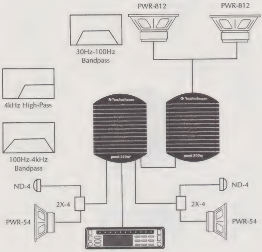


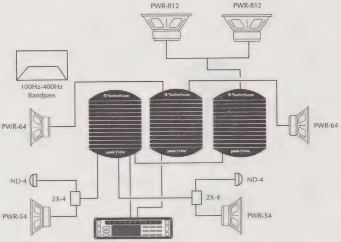
Crossover Points Added: 4kHz High/Low-Pass
Products Added: RFA-414
LP-1
System Configuration: 414 in Parallel with
Front Two Channels



Crossover Points Added: 30Hz High-Pass 60ix
100Hz Low-Pass 4080
Products Added: Punch 60ix
System Configuration: 4080 4-Channel Mode Fade Front to Rear with
the Summed Output Connected to 60ix

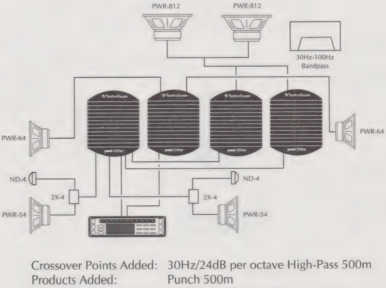
System Design





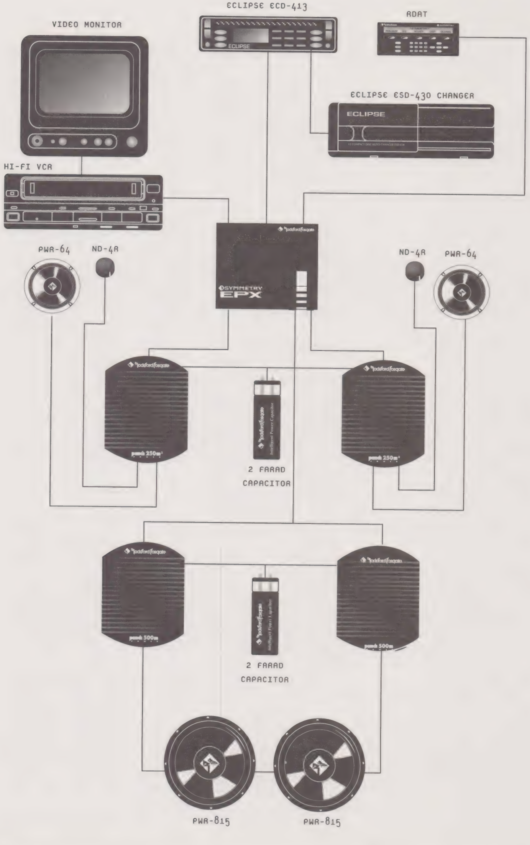
Crossover Points: (100Hz Low-Pass
100Hz High-Pass 250m^2)
30Hz High-Pass 500m
4kHz High/Low-Pass 2X-4
Products Used: Punch 250m^2/Punch 500m
2X-4/ND-4
PWR-54/PWR-812
System Configuration: 3-Way System Biamp with
Subsonic Filter on 500m

Crossover Points Added: 100Hz-400Hz Bandpass
Products Added: 250m^2
PWR-64
XM400
System Configuration: 4-Way with Electronic
Bandpass for Midbass



Crossover Points Added: 30Hz/24dB per octave High-Pass 500m
Products Added: Punch 500m
System Configuration: (2) 500m Bridged Mono
with 24dB/octave subsonic filter

1994 chevrolet Impala ss



Chapter 2

Amplifiers

- Creation & Construction
- Defining an Amplifier
- Design Features
- PowerCube
- PVC (Punch Verification Certificate)
- Series 1 Amplifiers
- Punch Amplifiers
- Amplifier Quick Reference
- Amplifier Accessories

AMPLIFIERS

INTRODUCTION

Rockford Fosgate is the leader in car audio amplifier manufacturing. We have invested over 2 million dollars in the most state-of-the-art manufacturing equipment and adopted a new production cell process. Our commitment in delivering the highest quality products is a direct representation of our manufacturing capability.

THE CREATION OF AN AMPLIFIER

The first process in the birth of a new amplifier is writing a charter document explaining the features, construction, and specifications of the amplifier by the New Product Development department. This charter document has a time line depicting when the product should be produced. The document is reviewed by the Marketing, Sales and Engineering departments. These departments then meet with the financial department to discuss viability of the product. Once approved, a team is formed and given the green light to proceed with the development of the product. The Engineering department proceeds with a board layout and submits it to the product team. The team reviews the board layout to make sure that it follows the charter document guidelines.

CONSTRUCTION STAGES

- ALPHA
- BETA
- PRODUCTION

There are three stages that a new product goes through: Alpha, Beta and Production. In the Alpha stage, the Engineering department hand-builds the product, and sends it to the team which has it tested by Production, New Product Development and RTTI. These three departments record their findings and present this information to the team.

The Beta stage is when the amplifier makes its first production run and is retested by New Product Development and RTTI. The amplifier is sent to predetermined Beta sites in the field with a preliminary owner's manual. The Beta sites put the product through normal retail conditions recording their finding and then submit it to New Product Development. The amplifier team reviews all of the data and makes the necessary changes. The final owner's manual is reviewed and approved and sent to be printed.

The production stage starts with the PC board in the surface mount room where the amplifier has surface mount components inserted into the board. It then goes through assembly stages where large through-hole parts are installed (ie: toroid coils, capacitors). The amplifier parts are soldered through the wave-solder machine. The quality control department reviews the product to ensure that it was assembled correctly by doing an initial test on it. The PC board is mounted to the heatsink and has a burn-in process that consists of 2kHz sine wave at 500mV. The amplifier also has its outputs shorted and runs into its thermal limits to ensure all protection circuits are fully operational. The final test is the Punch Verification Certificate that utilizes a 1kHz sine wave that is ramped up into steps and the output power is into a two Ohm resistive load. The amplifier is packed and sent to the warehouse for shipping.

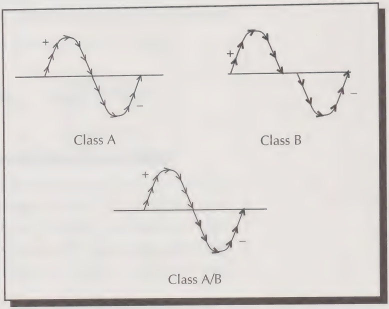
Defining An Amplifier

- Efficiency
- Low Distortion

An amplifier performs amplification or increases the amplitude of a signal. A car audio amplifier is an AC generator that converts DC voltage input to AC output voltage. In a perfect amplifier the amplifier would reproduce the same wattage output relative to the input wattage. For example, 100 Watts in equal 100 Watts out hence a 100% efficient amplifier. And the output signal should be identical to the input signal except for the change in amplitude. Any change to the signal is a result of distortion from the amplifier. A 100% efficient amplifier does not exist in the car audio domain. The average efficiency of a Rockford Fosgate amplifier is 60% which would represent a figure of 100 Watts in equal 60 Watts out.

Amplifiers are divided into classes that define their operating characteristics. There are 8 classes but in car audio amplification we deal with mainly two class; A and B. Class A amplifier is biased so it conducts the entire input cycle. This gives you high linearity (low distortion) because of the use of one device but low efficiency. In a Class A amplifier the efficiency can be as low as 25%. This is not useful in car audio when you have a limited current supply.

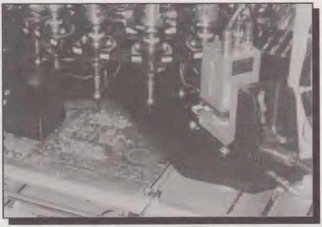
Class B conducts for just half of each input cycle and rests the remainder of the time. This allows this class to be 80% efficient at maximum power, but there is a need to use two transistors in a push/pull configuration to complete the full cycle. This can cause a non-linearity between the two transistors because the two halves of each output cycle are being amplified by two separate transistors. Class AB is a compromise and is widely used in car audio. It gains linearity of Class A and efficiency of Class B. This class reduces crossover distortion and increases the overall sound quality.



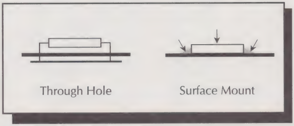
Discrete Surface Mount (DSM)

- Lower Operating Temperature
- Increased Efficiency
- Increased Reliability

DSM (Discrete Surface Mount) is an amplifier manufacturing process that utilizes a state-of-the-art pick and place machine which checks the tolerance and value of the discrete components. Rockford has a surface mount room with three surface mount machines. These machines have a combined capability to apply 20,000 components per hour. The Punch 40i has 350 components and will only take two minutes to insert all of the components.



These discrete components are applied to the surface of the PC board as opposed to through-hole components.

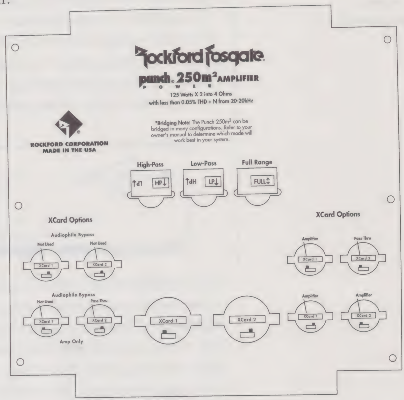


The discrete components are laser etched to meet a particular value and a very low tolerance. This lower tolerance improves the efficiency of the circuit, decreases crosstalk from other discrete components and lowers operating temperature. These superior components are used by aerospace, automotive, cellular, and mobile security manufacturers.

Electronic Crossover ("X")

- More efficient signal process
- Low Signal Loss
- Frequency Adjustable

The crossover circuitry is unique only to Rockford. It yields a wide ability in changing crossover points and "Q" factor. The crossover card is preset to full range internally in the amplifier and can be selected to be in high or low-pass at a 100Hz 12dB/octave slope. By using XCards, you can customize the crossover points to fit the need of the system. The placement of the speakers can be optimized by changing the "Q" of the crossover point. Positioning the crossover internally in the amplifiers enable utilizing them as a subsonic filter.



TRANS•NOVA

- **INCREASED SOUND QUALITY**
- **GREATER EFFICIENCY**
- **HIGHER RELIABILITY**

Trans•*nova* (**TRANS**conductance **NO**dal **V**oltage **A**mplifier) is a patented circuit that allows the audio signal to pass through the amplifier at low voltage. The trans•*nova* circuit was made possible by the advent of a new component in the car audio industry called a FET (Field Effect Transistor). These new amplifying devices became available around 1980 coming closer to the designer's ideal specifications than would ever have been expected. The new FETs encompass several technologies including vertical and lateral power MOSFETs and high gain, low noise junction FETs. These new devices have a close resemblance to the best of tube technology. Since these FETs behaved similarly to tubes, they provided an opportunity for creative circuitry with possibilities for product excellence, reliability and value. Thus began the developments leading to the trans•*nova* amplification circuitry.

TRANS•NOVA CIRCUITRY

The resulting design utilizes an output stage with a simpler gain structure and a shorter total signal path than conventional high voltage bi-polar driver (amplifier) designs. The number of stages is reduced from five or more to three. The output stage is further refined into a trans-impedance stage (current to voltage converter) to achieve a short loop (fast) negative feedback. The output stage is driven cooperatively by a transconductance stage (voltage to current converter).

TOPAZ

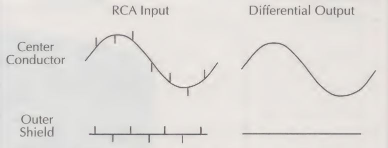
- **Increased Noise Rejection**
- **Lower Chance Ground Loops with the Source**

TOPAZ (Tracking Operation Pre-Amplifier Zone) circuitry solves ground loop noise problems common to automotive amplifier design. This innovative new development allows vastly improved isolation of the input signal grounds from the power supply ground of the amplifier. This is accomplished by allowing the source unit to control the potential "environment" of the entire input structure or "zone" of the amplifier. This process improves the noise rejection of the amplifier by 30-40dB; an astounding 20-100 times better than amplifiers without TOPAZ.

Balanced Line Inputs

- **Rejects Noise Pickup by the Cable**
- **Establishes a Stable Ground Point**

Using a BLT (Balanced Line Transmitter) at the source unit and balanced line inputs on the amplifier is the answer to rejecting any type of noise that will be introduced into the signal cable. The differential input circuitry used in the balanced input system rejects whatever signals are common to both of the shield twisted pair conductors. Balanced line is universal in concert installations where the stage and mixing consoles are hundreds of feet apart.



Differential inputs only amplify the difference between two conductors. Note: The noise spikes appear on both the center conductor and the outer shield and therefore are not amplified.

RTP NOMAD

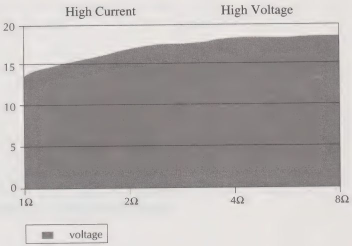
- **Non-Current Limiting**
- **Constant Amplifier Monitoring**
- **Does not Degrade Sound Quality**

The RTP NOMAD (Real Time Protection with NOn-Multiplying Advanced Decision) system uses an analog computer to make advanced decisions in real time based on device voltages to precisely control the awesome power available from the MOSFET output devices. This is the most sophisticated version of this technique ever used, bringing previously unavailable levels of accuracy, stability, temperature immunity and reliability to this critical process without limiting the current that the Rockford Fosgate amplifiers are famous for.

Control Rail Voltage (CRV)

- **Maximizes its Power into an Impedance Load**
- **High Voltage / High Current Capability**

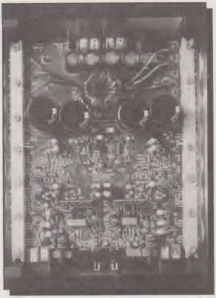
CRV (Control Rail Voltage) is designed to constantly maintain high power to all different impedances. This will enable a higher efficiency because the voltage and current threshold will compensate relative to the impedance it will see. This is an advantage over high current amplifiers that are only optimized for lower impedances and amplifiers with predetermined output voltage ratings that will turn off prematurely into lower impedances.



ITS

- **Increased Power Supply Efficiency**

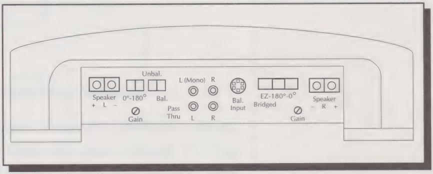
ITS (Increased Thermal Stability) power supply design is new in the $250m^2$ and 500m. This new toroidal power transformer design carries the high current input leads directly to the switching power MOSFETS. This both minimizes PC board heat and takes advantage of the natural air cooling of the leads.



Pass-Thru Circuit

- **System Flexibility**
- **Daisy Chaining an Additional amplifier**

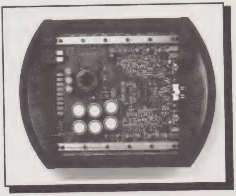
The Pass-Thru provides a convenient source for daisy chaining an additional amplifier. This eliminates need for additional RCA cable or "Y" adapters. One of the internal crossover can be designated to the Pass Thru output creating a dedicated low-pass, high-pass or full range output.



"I" Series

- **Increased Signal-to-Noise Ratio**
- **Less Connections**
- **Shorter Signal Path**

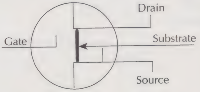
The use of surface mount technology enables 98% discrete components to be mounted directly to the PC board. Because of the shorter signal paths the signal-to-noise ratio and circuit reliability is increased. This new circuit design is unique to DSM amplifiers.



MOSFET Devices

- **Fast Switching**
- **Thermal Stability**

MOSFETS (Metal Oxide Semiconductor Field Effect Transistor) offer several important inherent advantages over bipolar design. These advantages are: thermal stability, switching speed, ultra low output impedance and wider bandwidth linearity. The devices are used in the power supply and output side and assist in delivering midrange clarity and low-end response.



N-Type

The gate controls the amount of current that can flow from the source to the drain. If higher negative voltage is applied to the gate, it will lower the amount of current from the source to the drain.

Cast Aluminum Heatsink

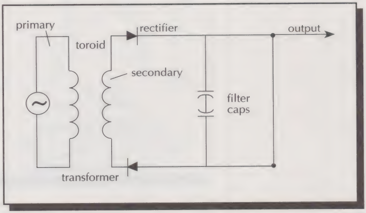
- Higher Transfer of Heat
- Increased Thermal Time

The heatsink is designed to absorb heat generated from the internal circuitry on the PC board. The cast heatsinks are finned for maximum surface area. All Punch amplifiers utilize the cast aluminum heatsink which has 30% more thermal time than an aluminum extrusion.

Loosely Regulated Power Supplies

- Increased Voltage and Current
- Lower Distortion

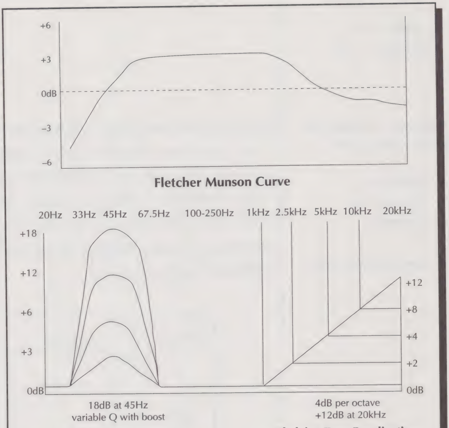
Power supplies step up the DC voltage and current into the amplifier by the use of a transformer. The DC voltage is converted to AC voltage then filtered to the switching devices. All Rockford amplifiers employ the use of pulse width modulation in the power supply. This circuit varies the pulse width of the DC to DC converter drive that will control the AC signal and the amount of voltage produced by the power supply. This will increase efficiency and dynamic range.



Punch Equalization

- Increased Bass Output
- Extended High Frequencies

The Punch Equalizer was designed to compensate for the uneven frequency response that people hear. The Fletcher Munson curve depicts the response that we perceive the level of certain frequencies to be. The Punch EQ circuit offers two types of equalization and over 5 bands of boost.



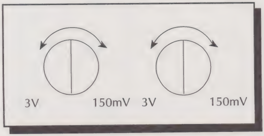
Peak Boost Equalization with increased Q relative to level. Qs equal to the center frequency divided by the bandwidth.

Shelving Type Equalization Boost increase per octave from the fundamental frequency.

Independent Gains

- Equal LR Outputs
- Able to Adjust an Imbalanced Signal from a Source

The Independent Input Gain controls are unique to all the Punch 2-channel amplifiers. This feature enables the amplifier to have the ability to correct an unequal level from any source unit.



The AudioGraph / PowerCube

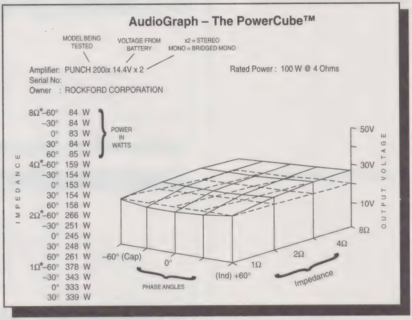
"Why Do We Test This Way?"

Amplifier testing has been used to measure the capabilities of amplifiers. The institute of high fidelity (IHF 202) test is recognized by manufacturers and is considered the industry standard in power ratings and has been used in car audio for numerous years.

The IHF-202 test consists of a 1kHz bursted tone which is pulsed on for 20 ms and then attenuated 20dB for 480 ms at a constant voltage (14.4 volts).

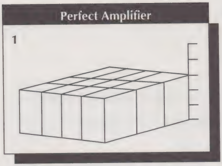
Some manufacturers give high power ratings but also have a high distortion rating sometimes 5% or higher total harmonic distortion. The power ratings given are from a purely resistive load. This is not the way an amplifier deals with music. Music is dynamic and is constantly changing.

The AudioGraph/PowerCube is a test instrument from Sweden that Rockford has utilized to establish power ratings that illustrate how an amplifier reacts to complex signals and inductive and capacitive loads of a speaker. The PowerCube is used to show that all amplifiers are not created equal. The PowerCube follows the same bursted tone at 1kHz with 20 ms on and 480 ms attenuated to 20dB. The 1kHz sinewave has .1% THD and the level is variable from 0 volts to 4 volts with 1 millivolt steps.



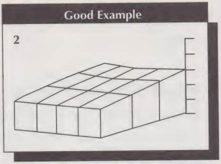
The maximum dynamic power is plotted on the graph and has a maximum distortion of 1%. Rockford tests their amplifiers in stereo and in mono at 14.4 volts and 12.6 volts. This can illustrate an ideal voltage and the static voltage of a vehicle's battery with the engine not running.

PERFECT AMPLIFIER



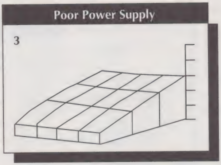
This amplifier works as a perfect voltage generator and is not affected by the connected load.

GOOD EXAMPLE



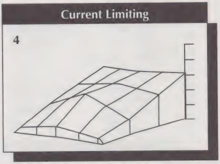
This is an amplifier with typically good behavior. There are some low impedances losses, but the PowerCube is still almost cubic.

POOR POWER SUPPLY



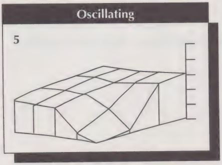
This amplifier's power supply is too small or has too few output devices and cannot produce enough current in low impedances. As a result, the dynamic output power gets lower in these impedances.

CURRENT LIMITING



This amplifier probably has too few output devices. To protect them, the amplifier has an electronic protection circuit which turns off the amplifier at "dangerous loads."

OSCILLATING



This amplifier is not stable at certain reactive loads. As a result, the distortion is always over 1% THD and the resulting output power is 0 watts.

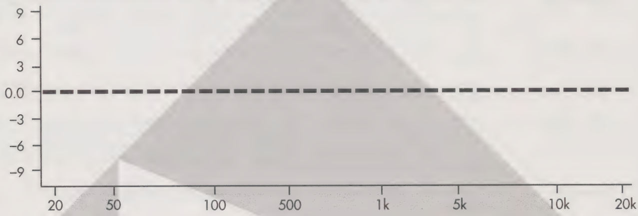
Rockford Fosgate

Punch Verification Certificate

Date: 01-05-95
Model: Punch 200ix
Serial #: B1THA4A00000003

Test Impedance: 2 Ohms
Test Voltage: 14.2 Volts DC

ACTUAL FREQUENCY RESPONSE PLOT



ACTUAL TEST MEASUREMENTS

TEST SEQUENCE	TEST PARAMETERS MIN	MAX	ACTUAL	
Idle Current	0.15	5.0	1.22	Amps Idle
Loaded Current	35.0	65.0	55.0	Amps Draw
Turn-on Delay	1.0	3.0	1.0	Seconds
Efficiency	40.0	—	76.9	Percentage
Output Voltage	162.0	—	24.5	Volts RMS
Output Power	—	0.8	300.5	Watts/Channel
Tracking Error		—	0.2	dB
Signal to Noise		—	>100dB	

TEST RESULTS PASSED

Punch Verification Certificate

Explanation of Test Parameters

Date:	The date recorded here is the day the amplifier was final tested and packed.
Model & Serial Number:	Every amplifier is tested. This assures Rockford Fosgate 100% quality control. Other manufacturers' amplifiers are "batch tested;" 1 of 100 or 1 of 1,000.
Test Impedance / Test Voltage:	A system designed with 4Ω and 8Ω midrange/tweeters and a 4Ω subwoofer system will present a 2Ω load to the amplifier. The *Test Voltage* is representative of what a properly working automotive charging system will provide the amplifier, allowing it to make its rated *Output Power. Watts per Channel = Volts^2 / Impedance.*
Actual Frequency Response Plot:	This test shows the exact frequency response of each amplifier. Notice that the amplifier is tested with the Patented Punch EQ at "0" effect to show the amplifier's audiophile type frequency response.

Actual Test Measurements - ***Minimum/Maximum/Actual***

Idle Current:	The amount of current the amplifier is drawing from the car without any music playing.
Loaded Current:	This test shows how much power this amplifier will draw with music playing. It is always relative to impedance and voltage. With the same voltage in a 4Ω application this amplifier will draw half the current it will at 2Ω. This test is very important when trying to determine the total amperage draw of a multi-amp system.
Turn-On Delay:	How many seconds before the amplifier un-mutes itself. This circuit protects the amplifier against turn-on and turn-off pops.
Efficiency:	Conversion efficiency, or the ability of the amplifier to convert input power to output power. The average in most amplifiers is 40-50%. The remaining % turns into heat.
Output Voltage / Output Power:	Output voltage and impedance (2Ω in this test) are used to calculate Output Power (watts per channel). RMS power is what the amplifier will do all the time, even under the worst conditions. Dynamic power is what the amplifier has on "Reserve." Rockford amplifiers will DOUBLE their rated output (even in MONO) for short periods. This gives you +3dB of Dynamic head room. This is why Rockford Fosgate amplifiers are rated so conservatively. Plain and simple you get more for your money.
Tracking Error:	Each channel is tested separately. They must test to within 0.8dB. If not, the channels would sound uneven or out of balance.
Channel Separation:	Sometimes called Crosstalk, is a measure of the isolation of one channel from another. The spec tells you how many dB of separation the channels operate with.
Signal to Noise:	Internal quietness of the amplifier. Music vs. noise. The larger the number the better.

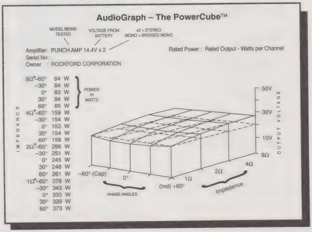
PowerCube™ by AudioGraph

which appears on the back of the amplifier box

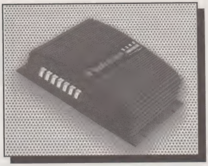
The PowerCube™ is a test instrument that measures the dynamic output power of an amplifier. It measures output voltage into an 8Ω, 4Ω, 2Ω and 1Ω load. Each load is also measured at –60°, –30°, 0°, +30° and +60° phase angle. This range of impedances and phase angles approximates the load that a typical three-way speaker system presents to the amplifier. A loud speaker is not a static or fixed load, its impedance and the phase relationship between the voltage and current will change based on the frequency of the signal passing through it. The measured information is displayed in the form of a cube. The Output Voltage is displayed on the vertical axis, the Phase is displayed on the horizontal axis and the Impedance is displayed on the Z axis.

A well-designed amplifier will produce the same amount of output voltage regardless of the load (impedance and phase) to which it is attached. This will ensure that the amplifier will make its rated power without clipping. Any output above the rated output is a bonus or headroom. The dotted line on the cube represents the rated output of the amplifier, the solid line is the actual measured output. Amplifiers of good design and honest power ratings will *fill up* the cube. Any additional output can be considered added value. If an amplifier produces less output at any particular impedance or phase angle, it will clip or distort when a signal excites the speaker to that impedance and phase angle.

The PowerCube™ also displays the dynamic output power (watts) at each load.



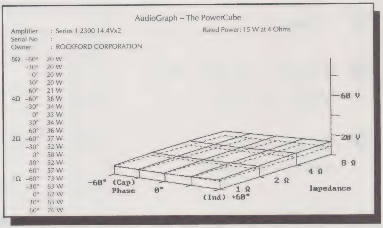
Series 1
2300

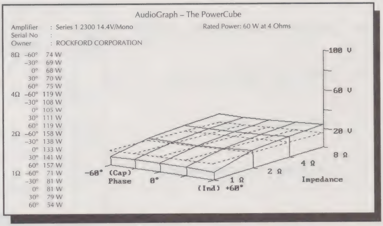


- 15 Watts x 2 into 4Ω .08% THD
- 30 Watts x 2 into 2Ω .30% THD
- 60 Watts Mono into 4Ω .30% THD

EXCLUSIVE ROCKFORD FEATURE

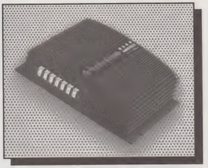
- "I" Series
- DSM Discrete Surface Mount Component
- RTP Real Time Protection Circuit
- 3 Year Warranty
- Terminal Connectors
- High Level Input
- Gold-Plated Connectors





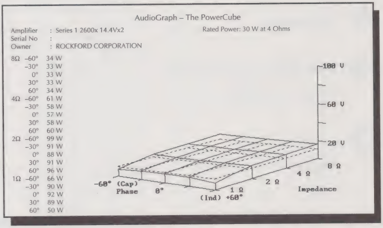
Series 1
2600X

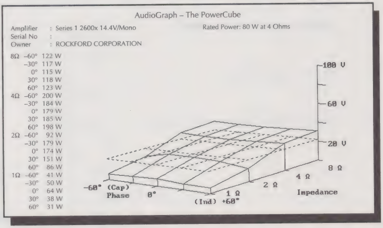
- 30 Watts x 2 into 4Ω .08% THD
- 60 Watts x 2 into 2Ω .30% THD
- 120 Watts Mono into 4Ω .30% THD



EXCLUSIVE ROCKFORD FEATURE

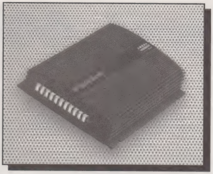
- "I" Series
- DSM Discrete Surface Mount Component
- RTP Real Time Protection Circuit
- 3 Year Warranty
- Electronic Crossover ("X")
- Terminal Connectors
- High Level Input
- Gold-Plated Connectors





Series 1

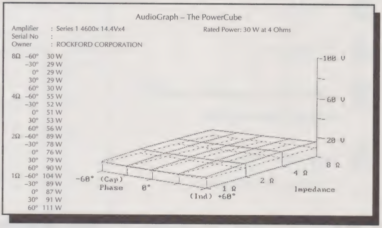
4600X



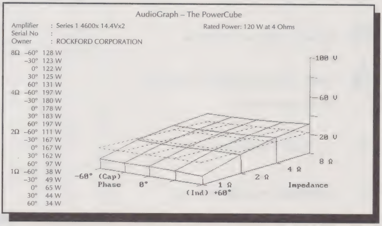
- 30 Watts x 4 into 4Ω .08% THD
- 60 Watts x 4 into 2Ω .30% THD
- 120 Watts x 2 into 4Ω .30% THD

Exclusive Rockford Feature

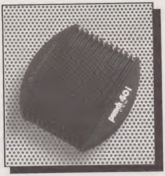
- "I" Series
- DSM Discrete Surface Mount Component
- RTP Real Time Protection Circuit



- 3 Year Warranty
- Electronic Crossover ("X") Front and Rear
- Terminal Connectors
- High Level Input
- Gold-Plated Connectors



punch® 40i DSM

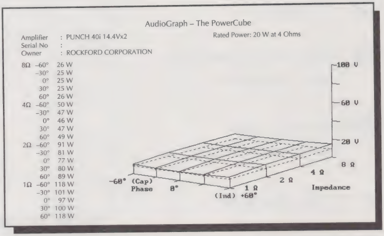


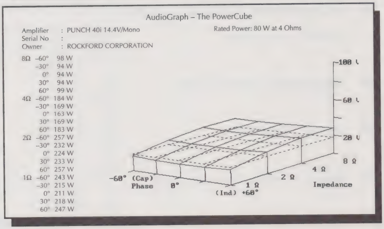
- 20 Watts x 2 into 4Ω .05% THD
- 40 Watts x 2 into 2Ω .1% THD
- 80 Watts mono into 4Ω .1% THD

EXCLUSIVE ROCKFORD FEATURE

"NEW"

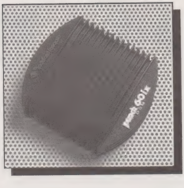
- TOPAZ
- "I" Series
- DSM 98% Discrete Surface Mount Component
- RTP Real Time Protection Circuit
- CRV Control Rail Voltage
- Punch Equalization
- Cast Aluminum Heatsink with End Caps





- 3 Year Warranty
- MOSFET Pulse Width Modulated Power Supplies
- Terminal Connectors
- Double-Sided Plated Through Glass Epoxy PC Board

punch® 60ix DSM

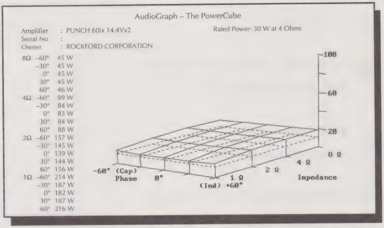


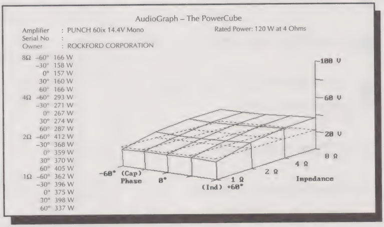
- 30 WATTS X 2 INTO 4Ω .05% THD
- 60 WATTS X 2 INTO 2Ω .1% THD
- 120 WATTS MONO INTO 4Ω .1% THD

EXCLUSIVE ROCKFORD FEATURE

"NEW"

- TOPAZ
- "I" SERIES
- DSM 98% DISCRETE SURFACE MOUNT COMPONENT
- RTP REAL TIME PROTECTION CIRCUIT
- CRV CONTROL RAIL VOLTAGE
- PUNCH EQUALIZATION
- ELECTRONIC CROSSOVER ("X")
- CAST ALUMINUM HEATSINK WITH END CAPS

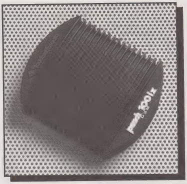




- MOSFET PULSE WIDTH MODULATED POWER SUPPLIES
- TERMINAL CONNECTORS
- DOUBLE-SIDED PLATED THROUGH GLASS EPOXY PC BOARD

punch® 100ix
DSM

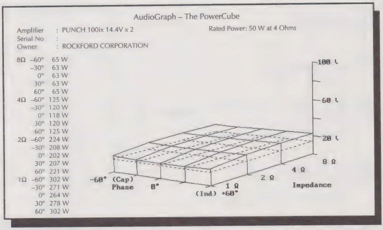
- 50 Watts x 2 into 4Ω .05% THD
- 100 Watts x 2 into 2Ω .1% THD
- 200 Watts mono into 4Ω .1% THD

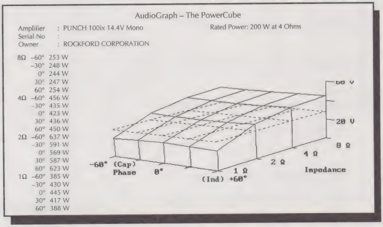


EXCLUSIVE ROCKFORD FEATURE

"NEW"

- TOPAZ
- "I" Series
- DSM 98% Discrete Surface Mount Component
- RTP Real Time Protection Circuit
- CRV Control Rail Voltage
- Punch Equalization
- Electronic Crossover ("X")
- Cast Aluminum Heatsink with End Caps

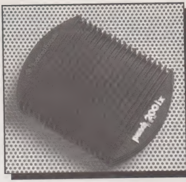




- 3 Year Warranty
- MOSFET Pulse Width Modulated Power Supplies
- Terminal Connectors
- Double-Sided Plated Through Glass Epoxy PC Board

punch® 200ix
DSM

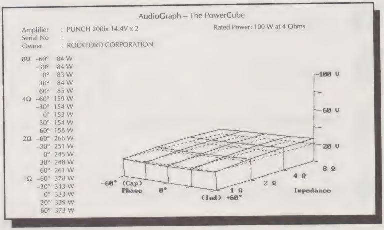
- 100 Watts x 2 into 4Ω .05% THD
- 200 Watts x 2 into 2Ω .1% THD
- 400 Watts mono into 4Ω .1% THD

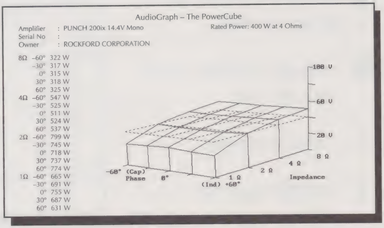


EXCLUSIVE ROCKFORD FEATURE

"NEW"

- TOPAZ
- "I" SERIES
- DSM 98% DISCRETE SURFACE MOUNT COMPONENT
- RTP REAL TIME PROTECTION CIRCUIT
- CRV CONTROL RAIL VOLTAGE
- PUNCH EQUALIZATION
- ELECTRONIC CROSSOVER ("X")
- CAST ALUMINUM HEATSINK WITH END CAPS





- 3 YEAR WARRANTY
- MOSFET PULSE WIDTH MODULATED POWER SUPPLIES
- TERMINAL CONNECTORS
- DOUBLE-SIDED PLATED THROUGH GLASS EPOXY PC BOARD

punch® 4020 DSM

- 20 Watts x 4 into 4Ω .05% THD
- 30 Watts x 4 into 2Ω .1% THD
- 60 Watts x 2 into 4Ω .1% THD

EXCLUSIVE ROCKFORD FEATURE

"NEW" TOPAZ

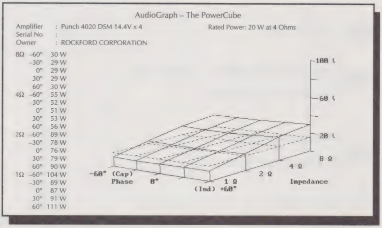
"NEW" RTP/NOMAD

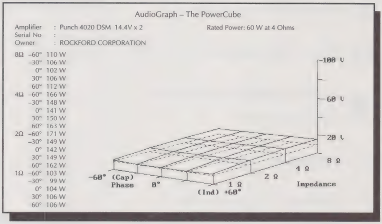
DSM 98% Discrete Surface Mount Component

CRV Control Rail Voltage

Electronic Crossover Front and Rear

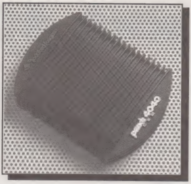
Cast Aluminum Heatsink with End Caps





- 3 Year Warranty
- MOSFET Pulse Width Modulated Power Supplies
- Double-Sided Plated Through Glass Expoxy PC Board

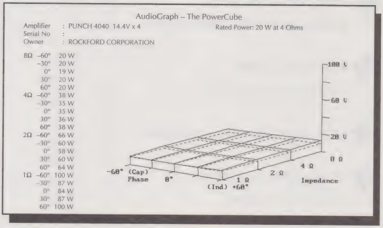
punch® 4040 DSM

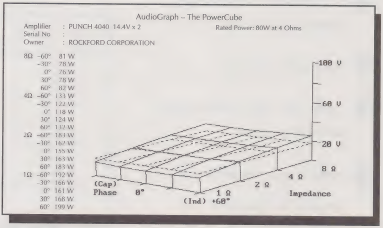


- 20 Watts x 4 into 4Ω .05% THD
- 40 Watts x 4 into 2Ω .1% THD
- 80 Watts x 2 into 4Ω .1% THD

Exclusive Rockford Feature

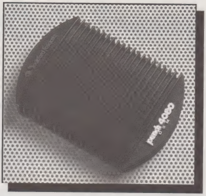
"NEW"
- Topaz
- DSM 98% Discrete Surface Mount Component
- RTP Real Time Proection Circuit
- CRV Control Rail Voltage
- Punch Equalization
- Electronic Crossover Front and Rear
- Polarity Switch
- Cast Aluminum Heatsink with end caps
- Pass-Thru With Xover
- 3 Year Warranty
- MOSFET Pulse Width Modulated Power Supplies
- Terminal Block Connectors
- Double-Sided Plated Through Glass Epoxy PC Board





punch® 4080 DSM

- 40 WATTS X 4 INTO 4Ω .05% THD
- 80 WATTS X 4 INTO 2Ω .1% THD
- 160 WATTS X 2 INTO 4Ω .1% THD



EXCLUSIVE ROCKFORD FEATURE

"NEW"

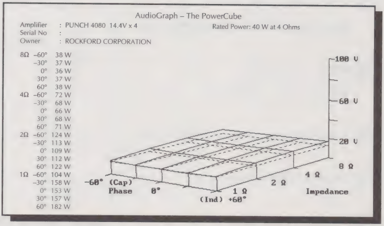
- TOPAZ
- DSM 98% DISCRETE SURFACE MOUNT COMPONENT
- RTP REAL TIME PROTECTION CIRCUIT
- CRV CONTROL RAIL VOLTAGE
- PUNCH EQUALIZATION FRONT AND REAR
- ELECTRONIC CROSSOVER FRONT AND REAR
- POLARITY SWITCH
- CAST ALUMINUM HEATSINK WITH END CAPS
- PASS-THRU WITH XOVER

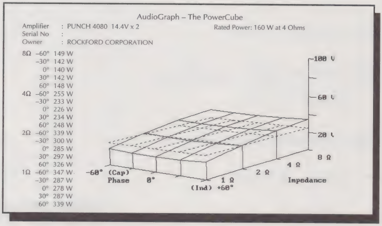
➻ 3 YEAR WARRANTY

➻ MOSFET PULSE WIDTH MODULATED POWER SUPPLIES

➻ TERMINAL BLOCK CONNECTORS

➻ DOUBLE-SIDED PLATED THROUGH GLASS EPOXY PC BOARD



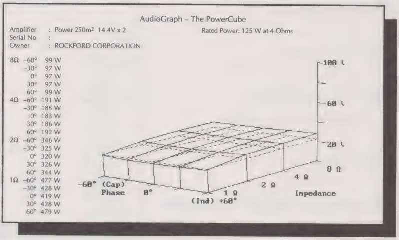


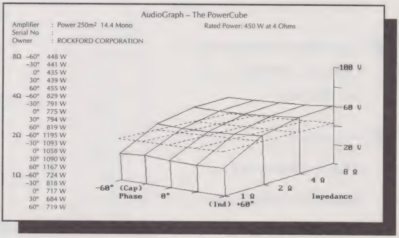
punch® 250m²
P O W E R

- 125 Watts x 2 into 4Ω .05% THD
- 225 Watts x 2 into 2Ω .1% THD
- 450 Watts mono into 4Ω .1% THD

EXCLUSIVE ROCKFORD FEATURE

- "NEW" TOPAZ
- "NEW" RTP NOMAD
- "NEW" TRANS•NOVA
- "NEW" BALANCED LINE INPUTS
- "NEW" ITS (INCREASED THERMAL STABILITY)
- DSM 98% DISCRETE SURFACE MOUNT COMPONENT
- "I" SERIES
- CRV CONTROL RAIL VOLTAGE
- ELECTRONIC CROSSOVER WITH 12DB AND 24DB/OCTAVE AND BYPASS
- RCA PASS-THRU JACKS





- 3 YEAR WARRANTY
- MOSFET PULSE WIDTH MODULATED POWER SUPPLIES
- DUAL PHASE SWITCH

punch® 500m
P O W E R

- 250 Watts x 1 into 4Ω .05% THD
- 500 Watts x 1 into 2Ω .1% THD
- 1000 Watts mono x 2 500m into 4Ω .1% THD

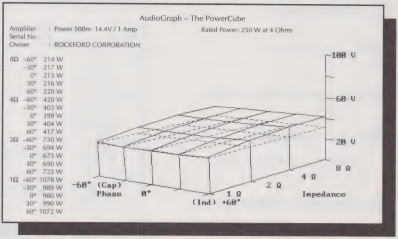
EXCLUSIVE ROCKFORD FEATURE

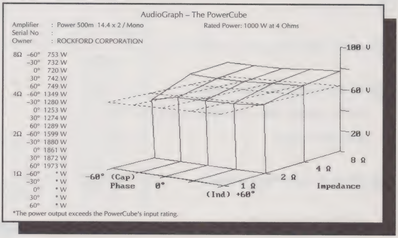
"NEW"
- TOPAZ

"NEW"
- RTP NOMAD

"NEW"
- Trans•nova

"NEW"
- ITS (Increased Thermal Stability)
- "I" Series
- CRV Control Rail Voltage
- DSM 98% Discrete Surface Mount Component
- Electronic Crossover 12dB and 24dB/octave and Bypass
- Pass-Thru
- 3 Year Warranty
- MOSFET Pulse Width Modulated Power Supplies
- Phase Switch

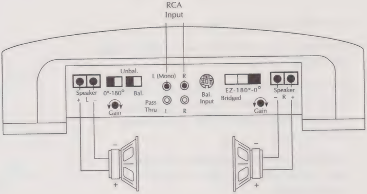




punch 250m²
P O W E R

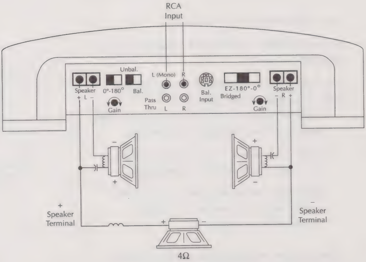
Stereo Operation

- Left channel phase at 0°.
- Signal input switch to unbalanced for RCA input.
- Right channel phase at 0° for stereo operation.
- Gain for left and right channels operate independently.
- XCard can be high-pass, low-pass or full range position.



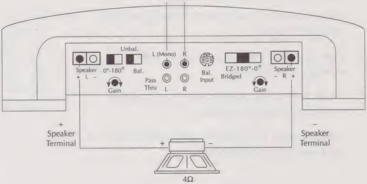
Stereo/Mono Operation

- Left channel phase at 0°.
- Signal input switch to unbalanced for RCA input.
- Right channel phase at 180° for stereo/mono operation.
- All speaker polarity on right channel is inverted to correct for signal phase.
- XCard is in full range position.



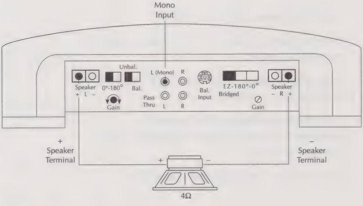
Bridged Operation

- Left channel phase at 0°.
- Signal input switch to unbalanced for RCA input.
- Setting the right channel phase at 180° will invert the signal and allow the subwoofer to operate correctly.
- Gain for left and right channels set equally to balance the subwoofer.
- XCard can be high-pass, low-pass or full range position.



E-Z Bridged Operation

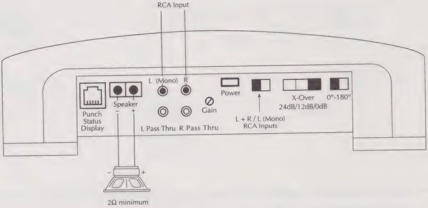
- Left channel phase at 0°.
- Signal input switch to unbalanced for RCA input.
- Selecting E-Z Bridged will allow the following to occur:
  - L (Mono) RCA input to drive both the left and right channels.
  - The left Gain to control both the left and right channels.
  - Inverts the right channel 180° allowing the subwoofer to operate correctly.
- XCard can be high-pass, low-pass or full range position.



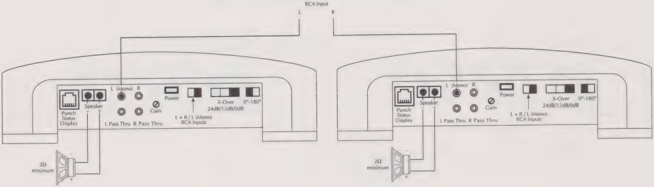
punch. 500m POWER

Mono Operation

- 500m set to L + R for RCA Input.
- Crossover set to 0dB, 12dB or 24dB.
- Phase at 0°.

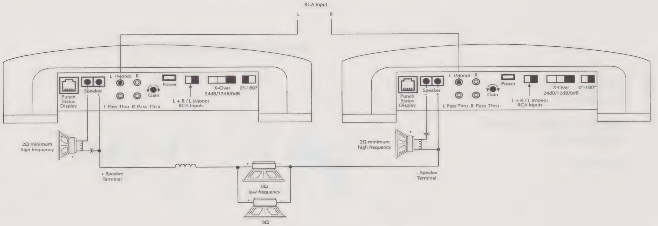


Stereo Operation



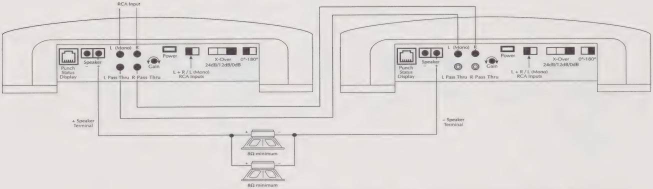
- Left and right amplifiers set to L mono for single RCA input.
- Crossover set to 0dB, 12dB or 24dB.
- Left and right amplifier phase at 0°.
- Gain for left and right amplifiers set equally to balance the subwoofer.

Stereo / Mono Operation



- Left and right amplifiers set to L (mono) for single RCA input.
- Crossover set to 0dB, 12dB or 24dB.
- Left amplifier phase at 0°.
- Right amplifier phase at 180°.
- All speaker polarity on right amplifier is inverted to correct for signal phase.
- Gain for left and right amplifiers set equally to balance the subwoofer.

Bridged Mono Operation



- Left and right amplifiers set to L + R for RCA input.
- Crossover set to 0dB, 12dB or 24dB.
- Left amplifier phase at 0°.
- Right amplifier phase at 180°.
- Gain for left and right amplifiers set equally to balance the subwoofer.

Amplifier Quick Reference

Each amplifier is two channels unless otherwise stated. Power rating is continuous RMS Watts per channel, from 20Hz to 20,000Hz, with all channels driven. Model numbers with the X suffix indicate XCard crossovers are included. Dimensions are given in inches (LxWxH). Metric dimensions are listed in italics and are given in centimeters.

Series 1: Includes battery fuse and fuseholder.

Model	Dimensions	Fuse/Type	Power RMS 4Ω	Power RMS 2Ω	Power Mono 4Ω
2300	$5\frac{1}{8}$ x $7\frac{3}{4}$ x 2 *13 x 20 x 5*	10A ATC	15 Watts	30 Watts	60 Watts
2600x	$6\frac{1}{8}$ x $7\frac{3}{4}$ x 2 *16 x 20 x 5*	20A ATC	30 Watts	60 Watts	120 Watts
4600x 4-Channel	$9\frac{1}{8}$ x $7\frac{3}{4}$ x 2 *23 x 20 x 6*	30A ATC	30 Watts	60 Watts	120 Watts

Punch 2-Channel: Includes Allen wrench and installation hardware, battery fuse and fuseholder, installation wire.

Model	Dimensions	Fuse/Type	Power RMS 4Ω	Power RMS 2Ω	Power Mono 4Ω
40i DSM	$9\frac{5}{8}$ x $9\frac{5}{8}$ x $2\frac{5}{8}$ *24 x 24 x 6*	20A ATC	20 Watts	40 Watts	80 Watts
60ix DSM	$10\frac{5}{8}$ x $9\frac{5}{8}$ x $2\frac{5}{8}$ *27 x 24 x 6*	30A ATC	30 Watts	60 Watts	120 Watts
100ix DSM	$11\frac{5}{8}$ x $9\frac{5}{8}$ x $2\frac{5}{8}$ *30 x 24 x 6*	40A ATC	50 Watts	100 Watts	200 Watts
200ix DSM	$12\frac{5}{8}$ x $9\frac{5}{8}$ x $2\frac{5}{8}$ *32 x 24 x 6*	50A AGU	100 Watts	200 Watts	400 Watts

PUNCH 4-CHANNEL: INCLUDES ALLEN WRENCH AND INSTALLATION HARDWARE, BATTERY FUSE AND FUSEHOLDER, INSTALLATION WIRE. (4020DSM DOES NOT INCLUDE INSTALLATION WIRE.)

Model	Dimensions	Fuse/Type	Power RMS 4Ω	Power RMS 2Ω	Power Mono 4Ω
4020 DSM	12 5/8 x 9 5/8 x 2 5/8 *32 x 24 x 6*	30A ATC	20 Watts	30 Watts	60 Watts
4040 DSM	12 5/8 x 9 5/8 x 2 5/8 *32 x 24 x 6*	30A ATC	20 Watts	40 Watts	80 Watts
4080 DSM	13 5/8 x 9 5/8 x 2 5/8 *35 x 24 x 6*	50A AGU	40 Watts	80 Watts	160 Watts

PUNCH POWER 250M²: WITH XCARD CROSSOVER, PHASE SWITCH. INCLUDES ALLEN WRENCH AND INSTALLATION HARDWARE. FUSE NOT INCLUDED.

Model	Dimensions	Fuse/Type	Power RMS 4Ω	Power RMS 2Ω	Power Mono 4Ω
250m²	13 5/8 x 9 5/8 x 2 5/8 *35 x 24 x 6*	60A AGU	125 Watts	225 Watts	450 Watts

PUNCH POWER 500M: MONO WITH XCARD CROSSOVER, PHASE SWITCH, TWIN POWER AND GROUND TERMINALS. INCLUDES ALLEN WRENCH AND INSTALLATION HARDWARE. FUSE NOT INCLUDED.

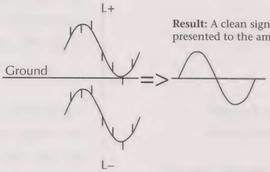
Model	Dimensions	Fuse/Type	Power RMS 4Ω	Power RMS 2Ω	Power Mono 4Ω
500m	13 5/8 x 9 5/8 x 2 5/8 *35 x 24 x 6*	60A AGU	250 Watts single amp	500 Watts single amp	1kWatt twin amp bridge

Amplifier Accessories

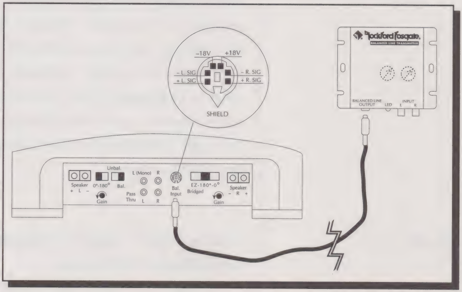
BLT

The BLT (Balanced Line Transmitter) sends voltage signals from the source unit to the amplifier. The BLT converts an RCA signal to a balanced line signal. The voltage is stepped up to 25 Volts RMS (12.5V each phase).

Differential inputs amplify the difference between two conductors. Any signal common to both channels will be subtracted from the final output.



Result: A clean signal is presented to the amplifier.



Punch Link

The Punch Link is a specially cast heatsink interconnect which allows you to join any of the current Punch or Punch Power amplifiers together.

Dimensions:
$2^{5}/_{8}$" H x $9^{5}/_{8}$" W x $3^{13}/_{16}$"L

PUNCH CAPACITORS

- **RP7405 / RP7410**

The Punch Energy Storage Capacitors are used to provide extra current needed by amplifiers to reproduce musical transients. The Punch capacitors have the natural ability to filter AC ripple caused by the alternator which can prevent potential noise. The Punch capacitors have 24kt. gold terminals and are available in .5 farad (RP7405) and 1 farad (RP7410) values.

INTELLIGENT CAPACITORS

- **RP7450 / RP7455**

The Intelligent Capacitors offer the same performance advantages as the Punch capacitors as well as incorporating smart electronics for protection and power monitoring. The smart electronic's circuitry offers reverse polarity protection as well as automatic charging when initially connected to the system. Voltage monitoring is accomplished with three LEDs for the 7450, while the 7455 uses a digital voltage readout. By monitoring the voltage, the circuitry allows the capacitor to remain active or be disengaged from the electrical system depending on the current demand needed from the amplifiers. The intelligent capacitors have 24kt. gold terminals and are available in 1 farad (RP7450) and 2 farad (RP7455) values.

XCARD

Additional cards are available for specific crossover points. These cards have Butterworth alignments.

XM00 =	XM50 = 50Hz	XM150 = 150Hz	XM400 = 400Hz
Blank	XM70 = 70Hz	XM200 = 200Hz	XM4.5kHz = 4,500Hz
Card	XM100 = 100Hz	XM275 = 275Hz	XM6.5kHz = 6,500Hz

CHAPTER 3
SPEAKERS

Speakers

Introduction

- **Rockford Acoustic Designs**

Rockford's passion for performance has grown since the acquisition of Carbonneau Speakers in 1986, now commonly known as Rockford Acoustic Designs (RAD). Since the acquisition, Rockford has offered a variety of product lines that satisfy many different applications and customer needs. The standards which Rockford uses to test speakers are distinguishable throughout the speaker industry and are often imitated by others. Rockford is one of the few who actually own their speaker manufacturing facility which allows for creative developments in product construction versus purchasing prefabricated "off-the-shelf" materials. To guarantee customer satisfaction, Rockford offers a generous 2 year warranty to insure the craftsmanship and materials used in the manufacturing of our speakers.

Testing for Specifications

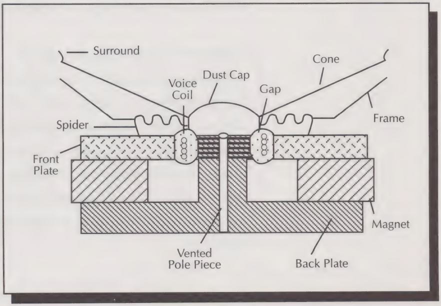
Rockford's procedure for testing speakers is adapted with the EIA 426 (Electronics Institute Association). This testing procedure is used to determine the **thermal** capacity of the speaker. This is done by suspending it free field (not a baffle board or enclosure) while applying a filtered white noise for 8 hours continuously. **Sensitivity** measurements for a sample driver are first calculated mathematically with the driver's Q-data, then verified near field and far field in real time with a microphone, a tone meter and using a frequency range that corresponds to piston diameter. The voltage applied is 2.00 volts for a 4 Ohm speaker and 2.83 volts for an 8 Ohm speaker.

Rockford's engineering design and use of materials exceed many manufacturer's best efforts. A major consideration when designing a driver is heat build-up. In conjunction with the piston movement a speaker naturally creates, heat transfer is aided with the use of a vented pole piece, aluminum voice coil former, and an expensive additive called ferrofluid. Durability of the speaker is accomplished with the use of cast aluminum frames, woven Kevlar™ cones and powder coat finishing. Last, but most importantly, sound quality is accomplished with the use of Nitrile rubber surrounds.

From Rockford's *Passion for Performance* to design standards, they don't call it ***The Punch*** for nothing!

Operation of a Speaker

Speakers are an electroacoustic device that convert electrical energy into sound energy and have an efficiency of .5% to 1%. Most speakers are based off a cone or dome that is put into motion by the reactance of AC voltage and a magnetic field. There are four parts that are comprised to make a speaker operate. The first is the motor which has a magnet, pole piece, front plate/gap and voice coil. The second is the diaphragm which is a cone with a dust cap or a solid dome. The third part is the suspension which has a spider and surround and the fourth is the frame. The motor is responsible for the movement of a speaker. A magnetic field is produced by a permanent magnet which is applied across the gap. The other element needed is AC current applied to the voice coil. The flow of current in one direction on the positive half of the cycle at a particular frequency causes the speaker to move into one direction. The reverse of the current on the negative half of the cycle again causes the speaker to change in direction. The cone is the critical factor in the sound quality because it vibrates the air which causes sound waves. If the cone body is too light, it can distort (or flex); if it is too heavy, the midrange response will suffer. The suspension is composed of the spider and surround and is responsible for centering the voice coil in the gap and the excursion of the speaker. The frame of the speaker is either stamped steel or cast aluminum and is responsible for the physical structure of the speaker.



Kevlar™ Reinforced Cone

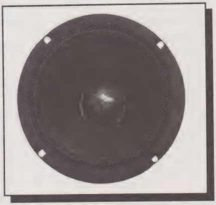
- Increased Cone Strength
- Accurate Low Frequency Reproduction

The Kevlar™ is mixed with the paper pulp in the construction of the cone. This additive adds strength to the body of the cone but does not increase weight. This type of cone is used in all midranges and woofers to increase frequency response.

Woven Kevlar™ Cone

- Increased Transient Response
- Low Cone Distortion

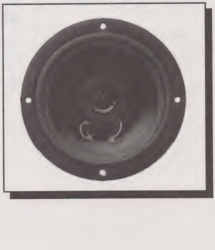
Woven Kevlar™ fibers are used in the Power Series to increase. Kevlar is used in bullet proof vests, helmets and in woven form in the Power Series Separates. The strength and light weight of Kevlar™ can lower the moving mass of the speaker and increase frequency transient response and lower cone distortion.



PVA Treated Cones

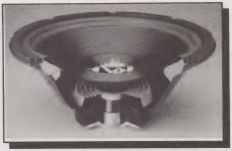
- Increases Cone Life
- Strengthen Cone Body

PolyVinyl Acetate is a coating that is applied to the speaker to prevent damage from UV rays. This will also increase the resistance to moisture and humidity so the paper cone does not break down from the automotive environment.



DUAL LAMINATED FOAM SURROUND

- **LOWERS EDGE RESONANCE**
- **PROVIDES SUSPENSION LINEARITY**



This half roll surround has very good termination quality at the cone. The curved shape of the roll surround enables it to have extra strength from the front and rear sides of the speaker under low frequency excursion. The dual laminate lowers the edge resonance which will decrease cancellation.

FLAT SPIDER DESIGN

- **LINEAR CONE CONTROL**
- **VOICE COIL CENTERED**

The main purpose of the spider is to assist in the control of the speaker and keep the voice centered in magnetic gap. The spider should also have the ability of handling excessive heat from the voice coil and not inhibit the excursion of the speaker.

NITRILE SURROUND

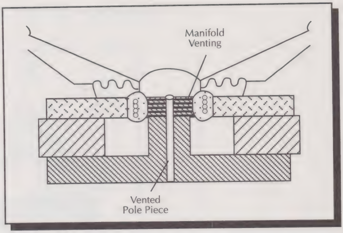
- **INCREASED EXCURSION**
- **BETTER CONE CONTROL**

The surround's purpose is to assist in the control of the speaker with the spider and provide the necessary excursion at low frequencies. The surround will absorb energy at the cone's edge to reduce any resonances. The Nitrile rubber surround enable the speaker to have an increased excursion and a more linear movement.

Vented Pole Piece

- Lowers Operating Temperature
- Increases Power Handling

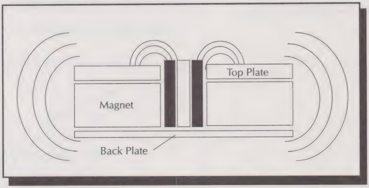
The main objective of the vented pole piece is to reduce the operating temperature of the voice coil. The pole piece allows fresh air to circulate inside the motor assembly. This will reduce the heat generated from the power being dissipated in the voice coil. The speaker's thermal power handling will increase and stabilize the impedance of the speaker.



Neodymium Magnet

- High Magnetic Field Strength
- Small in Size

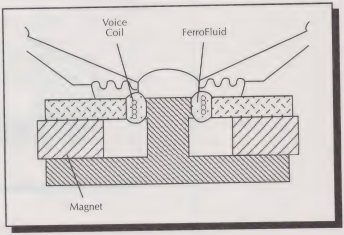
The Neodymium magnet has over **3 times** the flux density of a ferrite magnet. This enables the magnet to have one third the weight of a ferrite but have the same strength. The Neodymium magnet is utilized in the tweeter of the PCH-34, 44, 54, 64T2 and ND-4/8.



FERROFLUID

- **HIGH THERMAL CONDUCTIVITY**
- **LOWER IMPEDANCE RISE**

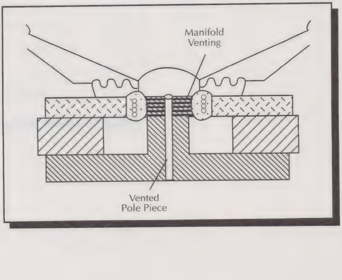
Ferrofluid is composed of a synthetic hydrocarbon, oil based fluid with billions of submicroscopic magnetic particles. This will act like a liquid bearing to lower the possibility of the voice coil rubbing. The fluid is injected into the gap where it is held by the magnetic field. Ferrofluid has four times the thermal conductivity as air. This will decrease frequency peaks and lower the impedance rise of the speaker.



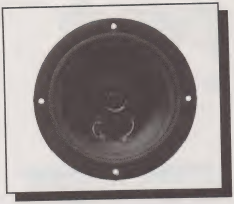
MANIFOLD VENTING

- **NO LOSS OF FERROFLUID**
- **CROSS VENTILATION**

Manifold Venting is a process where holes are drilled through the pole piece to relieve the pressure built up from the movement of the voice coil in the gap. This will also assist in the cooling of the voice coil along with keeping the Ferrofluid from escaping. This feature is included on speakers with Ferrofluid.



punch CLASSIC TWINS



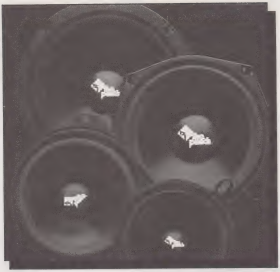
FEATURES

"NEW"
PCH-464T2 4" x 6"
"NEW"
PCH-54T2U 5.25"
"NEW"
PCH-4104T2 4" x 10"

- EURO DIN MOUNT ON 54T2, 44T2, 34T2
- KEVLAR™ REINFORCED CONES
- NEODYMIUM MAGNET ON 54T2U, 54T2D, 44T2, 34T2
- DUAL LAMINATE FOAM SURROUND
- LARGER VOICE COIL
  1" (34T2, 44T2, 54T2)
  1.25" (64T2)
- PVA TREATED CONES

punch

MIDRANGES



FEATURES

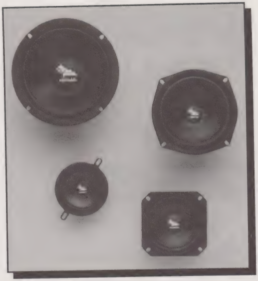
- STEP UP FEATURE
- FERROFLUID (ND-4, SP-34)
- NEODYMIUM MAGNET (ND4)
- KEVLAR™ REINFORCED CONES
- PVA TREATED CONES
- DUAL LAMINATE SURROUNDS
- VENTED POLE PIECE

punch SPLITS

FEATURES

- STEP UP FEATURE
- NITRILE RUBBER SURROUND
- FLAT SPIDER (PCH44/54)
- PCH-142X CROSSOVER
  PCH-14X WITH SPLITS SYSTEM
- VENTED POLE PIECE
- KEVLAR™ REINFORCED CONE
- PVA TREATED CONES

punch
AUDIOphile

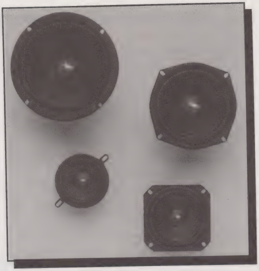


FEATURES

- STEP UP FEATURE

- NITRILE RUBBER SURROUND
- FLAT SPIDER
- VENTED POLE PIECE
- KEVLAR™ REINFORCED CONE
- PVA TREATED CONES
- 2X-4 CROSSOVER (RFA414/514/614)
  ND-4 WITH SYSTEM

punch
POWER



FEATURES

- STEP UP FEATURE
- WOVEN KEVLAR™ CONE
- FERROFLUID
- MANIFOLD VENTING
- LARGER VOICE COILS
  1" (PWR-34/44)
  1.5" (PWR-54/64)
- PVA TREATED CONES
- VENTED POLE PIECE

Speaker/Tweeter Quick Reference

Midrange Speakers: All midrange speakers come in pairs with mounting hardware, grilles, wire and manual. The Punch and Power Series are available in four and eight ohm versions; the Audiophile Series and Punch Splits are available in four ohms only.

Punch Midrange

Model	Frequency Response	Power Rating	Mounting Diameter	Mounting Depth
SP-64/68	44Hz-10kHz	80W	$5\frac{11}{16}$" (145cm)	$3\frac{3}{16}$" (81cm)
SP-54/58	96Hz-6kHz	50W	$4\frac{15}{16}$" (125cm)	$2\frac{3}{8}$" (60cm)
SP-44/48	85Hz-12kHz	50W	$3\frac{27}{32}$" (98cm)	$2\frac{3}{16}$" (56cm)
SP-34/38	205Hz-20kHz	50W	$3\frac{13}{32}$" (87cm)	$2\frac{3}{8}$" (60cm)

Audiophile Midrange

Model	Frequency Response	Power Rating	Mounting Diameter	Mounting Depth
RFA-64	65Hz-6kHz	80W	$5\frac{27}{32}$" (148cm)	$3\frac{3}{32}$" (79cm)
RFA-54	44Hz-6kHz	50W	$4\frac{15}{16}$" (125cm)	$2\frac{5}{8}$" (67cm)
RFA-44	94Hz-7kHz	30W	$3\frac{13}{16}$" (97cm)	$1\frac{27}{32}$" (47cm)
RFA-34	145Hz-10kHz	50W	$3\frac{13}{32}$" (87cm)	$1\frac{3}{8}$" (35cm)

Power Midrange

Model	Frequency Response	Power Rating	Mounting Diameter	Mounting Depth
PWR-64/68	65Hz-6kHz	120W	$5\frac{5}{8}$" (143cm)	$2\frac{31}{32}$" (75cm)
PWR-54/58	70Hz-7kHz	120W	$4\frac{7}{8}$" (124cm)	$2^{21}32$" (68cm)
PWR-44/48	85Hz-9kHz	100W	$3\frac{13}{16}$" (97cm)	$2\frac{5}{32}$" (55cm)
PWR-34/38	150Hz-10kHz	80W	$3\frac{13}{32}$" (87cm)	$1\frac{3}{8}$" (35cm)

Tweeters: All tweeters come packed in pairs with crossovers. The package contains speakers, crossovers and mounting hardware. All tweeter crossovers utilize the Optozorb protection system.

Tweeters

Model	Mounting	Crossover Model	Frequency/Slope	Power Rating
"NEW" PCH-14X "NEW" PCH-18X	Swivel Swivel	TX-1418 TX-1818	6kHz/18dB per octave	50W
SD4-X SD8-X	Fleximount Plate	TX-1418 TX-1818	6kHz/18dB per octave	50W
ND4-XFM ND8-XFM	Flush	TX-184N TX-188N	4kHz/18dB per octave	50W

Speaker Systems: All systems include a pair of midrange speakers and tweeters with two-way crossovers. Crossovers utilize Optozorb tweeter protection. Package includes speakers, crossovers, mounting hardware, grilles and installation manual. A mounting plate to fit a 4 x 6 cutout is included with the PCH-314.

Punch Splits

Model	Midrange	Tweeter	Crossover	Freq. Response	Power
"NEW" PCH-614	PCH-64	PCH-14	PCH-142X	45Hz-20kHz	50W
"NEW" PCH-514	PCH-54	PCH-14	PCH-142X	50Hz-20kHz	50W
"NEW" PCH-414	PCH-44	PCH-14	PCH-142X	100Hz-20kHz	50W
"NEW" PCH-314	PCH-34	PCH-14	TX-1418/ CC-100	225Hz-20kHz	50W

Audiophile

Model	Midrange	Tweeter	Crossover	Freq. Response	Power
RFA-614	RFA-64	ND-4	2X-4	29Hz-20kHz	80W
RFA-514	RFA-54	ND-4	2X-4	44Hz-20kHz	50W
RFA-414	RFA-44	ND-4	2X-4	94Hz-20kHz	30W

Twins: The Twins are two-way, full range, coaxial speakers. Package includes two speakers, manual, mounting hardware, grilles and wire (except PCH-34T2 which has no grilles). All power ratings are RMS continuous Watts. The Din Twins are standard sized stock replacement speakers for imports. Package contains speakers and manual.

Punch Twins

Model	Size	Mounting Depth	Frequency Response	Power Rating
PCH-694T2	6 x 9	$3\frac{9}{32}$" *(83cm)*	65Hz-20kHz	80W
"NEW" PCH-4104T2	4 x 10	$2\frac{15}{16}$" *(75cm)*	65Hz-20kHz	50W
PCH-64T2	$6\frac{1}{2}$	$2\frac{1}{8}$" *(54cm)*	80Hz-20kHz	80W
PCH-54T2	$5\frac{1}{4}$	$2\frac{5}{8}$" *(67cm)*	87Hz-20kHz	80W
"NEW" PCH-54T2U	$5\frac{1}{4}$	$1\frac{15}{16}$" *(49cm)*	88Hz-20kHz	50W
"NEW" PCH-464T2	4 x 6	$1\frac{29}{32}$" *(48cm)*	112Hz-20kHz	50W
PCH-44T2	4	$1\frac{3}{4}$" *(45cm)*	164Hz-20kHz	50W
PCH-34T2	$3\frac{1}{2}$	$1\frac{13}{32}$" *(36cm)*	175Hz-20kHz	50W

Punch Twins

Model	Size	Mounting Depth	Frequency Response	Power Rating
PCH-54T2D	$5\frac{1}{4}$ DIN	$1\frac{31}{32}$" *(50cm)*	87Hz-20kHz	50W
PCH-44T2D	4 DIN	$1\frac{5}{8}$" *(41cm)*	110Hz-20kHz	50W
PCH-34T2D	$3\frac{1}{2}$ DIN	$1\frac{1}{4}$" *(32cm)*	240Hz-20kHz	30W

Chapter 4
Woofers

- Enclosure Designs
- Design Features
- Series 1 Woofers
- Punch RF Woofers
- Punch Audiophile Woofers
- Punch Power Woofers
- Specifications
- Box That Rocks
- Aeroport Tube
- Nomogram
- Term-Pro

WOOFERS

ENCLOSURE DESIGNS

- **ACOUSTIC SUSPENSION**
- **BASS REFLEX**
- **SEALED BANDPASS**

The function of an enclosure is to assist the response of the woofer. Three common types of enclosures are acoustic suspension, bass reflex, and sealed bandpass. Not all woofers work very well in these three enclosures due to their inherent design. The enclosure must be designed around the woofer's parameters and the type of response you are trying to achieve.

An **acoustic suspension** enclosure (also known as a sealed box) is fairly simple to construct and behaves like an infinite baffle with additional stiffness caused by the air volume trapped in the enclosure is greater than outside air resistance which can contribute to good control of the woofer. Good power handling and transient response of the woofer is achieved in this enclosure. The sealed enclosure has a roll-off of 12dB/per octave from resonance. Generally, this enclosure provides control for high sound pressure levels and prominent bass impact.

The **bass reflex** enclosure is a closed box with a vent or port. This vented enclosure increases the efficiency of the low frequency response and increases the output by 3dB. The port provides the majority of the output at the tuned frequency of the enclosure. For best performance the port should be mounted on the same baffle with the woofer in order to take advantage of the operation of bass reflex design. A bass reflex enclosure exhibits a 24dB/octave cutoff from the tuned frequency. These enclosures have a lower transient response when compared to a sealed enclosure but better bass extension and some cases increased power handling.

A **sealed bandpass** enclosure is a combination of a sealed box and bass reflex. In a bandpass enclosure the woofer is enclosed in a closed box and fired into a bass reflex enclosure where the acoustic output then passes out through a port or vent. This design can have improved efficiency 3dB to 5dB in the bandwidth and offers and better low frequency extension with reduced output or a narrow bandwidth with increased output and better linearity. These three enclosures are fundamental in their design and some woofers will operate better than others. Only by deciding the response and output you are looking for will direct you to the proper enclosure. Utilizing a computer aided enclosure design program can illustrate what enclosure will work the best for the response that you need.

Inverted Dust Cap

- Improves Cone Rigidity
- Lowers Voice Coil Temperature

The inverted dust cap is designed to seal the woofer and provide protection to the voice coil gap. This will form an acoustic chamber which can generated air pressure across the former that can assist in the cooling of the voice coil.

Cast Aluminum Frame

- Less Frame Flex
- Rigid Stable Chassis

The cast aluminum frame offers increased stability over a stamped steel frame. This frame offers the lowest coloration to the sound because of the added strength of the frame to reduce any flexing.

Oversized Voice Coils

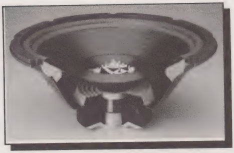
- Increased Power Handling
- Larger Cooling Area

The voice coil consists of wire wound on the aluminum, anodized former. The voice coil receives the AC current from the amplifier which causes the speaker to move away from the electromagnetic induction. The increased area of a voice coil can increase power handling because more windings of wire gives a better cross-sectional area to assist in cooling.

DUAL LAMINATED FOAM SURROUND

- LOWERS EDGE RESONANCE
- PROVIDES SUSPENSION LINEARITY

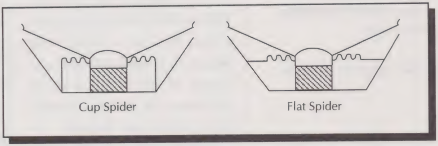
This half roll surround has very good termination quality at the cone. The curved shape of the roll surround enables it to have extra strength from the front and rear sides of the speaker under low frequency excursion. The dual laminate lowers the edge resonance which will decrease cancellation.



FLAT SPIDER DESIGN

- LINEAR CONE CONTROL
- VOICE COIL CENTERED

The main purpose of the spider is to assist in the control of the speaker and keep the voice centered in magnetic gap. The spider should also have the ability of handling excessive heat from the voice coil and not inhibit the excursion of the speaker.



NITRILE SURROUND

- INCREASED EXCURSION
- BETTER CONE CONTROL

The surround's purpose is to assist in the control of the speaker with the spider and provide the necessary excursion at low frequencies. The surround will absorb energy at the cone's edge to reduce any resonances. The Nitrile rubber surround enable the speaker to have an increased excursion and a more linear movement.

Kevlar™ Reinforced Cone

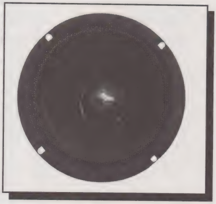
- Increased Cone Strength
- Accurate Low Frequency Reproduction

The Kevlar™ is mixed with the paper pulp in the construction of the cone. This additive adds strength to the body of the cone but does not increase weight. This type of cone is used in all midranges and woofers to increase frequency response.

Woven Kevlar™ Cone

- Increased Transient Response
- Low Cone Distortion

Woven Kevlar™ fibers are used in the Power Series to increase cone strength. Kevlar is used in bullet proof vests, helmets and in woven form in the Power Series Separates. The strength and light weight of the woven Kevlar™ can lower the moving mass of the speaker and increase frequency transient response and lower cone distortion.



PVA Treated Cones

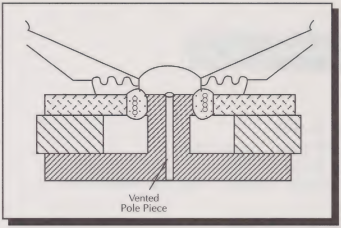
- Increases Cone Life
- Strengthen Cone Body

PolyVinyl Acetate is a coating that is applied to the speaker to prevent damage from UV rays. This will also increase the resistance to moisture and humidity so the paper cone does not break down from the automotive environment.

Vented Pole Piece

- Lowers Operating Temperature
- Increases Power Handling

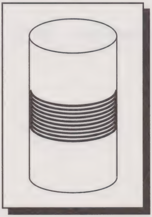
The main objective of the vented pole piece is to reduce the operating temperature of the voice coil. The pole piece allows fresh air to circulate inside the motor assembly. This will reduce the heat generated from the power being dissipated in the voice coil. The speaker's thermal power handling will increase and stabilize the impedance of the speaker.



Anodized Aluminum Former

- Increases Heat Dissipation

All Rockford Fosgate woofers have an Anodized Aluminum Voice Coil Former. This can assist in dissipating the heat generated by voltage through the voice coil and increase the performance of the woofer.



SERIES 1

FEATURES

- CRS BASKET
  HEAVY STEEL FOR RIGIDITY
  ANTI-CORROSION PAINT
  (ELECTROSTATICALLY APPLIED, BAKED ON)

"NEW"
- VENTED POLE PIECE

- PVA TREATED CONE

- ONE PIECE STEPPED BACK PLATE
  ALLOWS EXTENDED VOICE COIL EXCURSION
  VOICE COIL CAN'T BOTTOM OUT ON BACK PLATE

- ANODIZED ALUMINUM VOICE COIL FORMER

PUNCH®

FEATURES

STEP UP FEATURE

"NEW"

- INCREASED INVERTED DUST CAP SIZE
- KEVLAR® REINFORCED SPRUCE PULP CONE (KSP)
- DUAL LAMINATE SURROUND
- CHAMFERED FIELD PLATE
  TOP FIELD PLATE HAS BEVEL TO CREATE FOCUSED
  MAGNETIC IMAGE ON VOICE COIL
- LARGE STRONTIUM FERRITE MAGNET
  HIGH POWER HANDLING
  STACKED MAGNETS FOR IMPROVED SENSITIVITY (12", 15" AND 18")
- FLAT SPIDER (8", 10")
- ANODIZED ALUMINUM VOICE COIL FORMER
- VENTED POLE PIECE
- ONE PIECE STEPPED BACK PLATE
  VOICE COIL CAN'T BOTTOM OUT
  ALLOWS EXTENDED VOICE COIL EXCURSION

punch AUDIOphile

FEATURES

- STEP UP FEATURE

"NEW"
- INVERTED DUST CAP
- NITRILE RUBBER SURROUND
- LONGER VOICE COIL
  - INCREASED LINEAR EXCURSION (XMAX)
  - 2" VOICE COIL (8" AND 10")
  - 2.5" VOICE COIL (12")
  - INCREASED HEAT DISSIPATION
- FLAT SPIDERS
- KEVLAR® (KSP) CONE
- ANODIZED ALUMINUM VOICE COIL FORMER
- VENTED POLE PIECE
- LARGE STONTIUM FERRITE MAGNET
- STACKED MAGNETS FOR IMPROVED SENSITIVITY (10" AND 12")

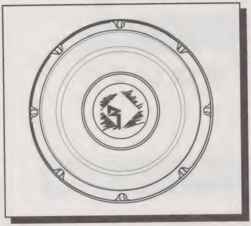
Punch POWER

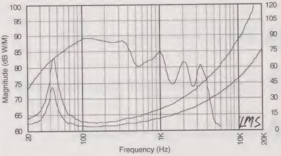
FEATURES

- STEP UP FEATURE
- WOVEN KEVLAR® CONE ON 8"
- THREE INCH AND FOUR INCH VOICE COILS
- CAST ALUMINUM BASKETS
- GOLD-PLATED BINDING POSTS
- MAGNET RUBBER BOOT
- ANODIZED ALUMINUM VOICE COIL FORMER
  CRS BASKET ON 8"
- VENTED POLE PIECE
- DUAL LAMINATED FOAM SURROUNDS
- FLAT SPIDERS

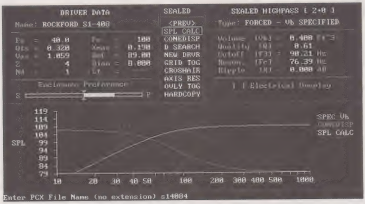
Woofer Specifications

	S1-408	S1-808
Freq. Range	40Hz-1,200Hz	42Hz-1,200Hz
Impedance	4Ω	8Ω
DCR	3.5Ω	6.9Ω
FS	40Hz	42Hz
QMS	4.02	2.90
QES	.35	.37
QTS	.32	.33
SD	34.10 in.^{2} .0.02220 m^2	34.10 in.^{2} 0.02220 m^2
VAS	1.059 ft.^{3} 30 L	1.059 ft.^{3} 30 L
Voice Coil Diameter	1.5 in. 38.1 mm	1.5 in. 38.1 mm
XMAX	.19 in. 4.7 mm	.19 in. 4.7 mm
Sensitivity	89dB W/m	89dB W/m
Power Handling	100 W	100 W
VD	.04 ft.^{3} 1.1 L	.04 ft.^{3} 1.1 L

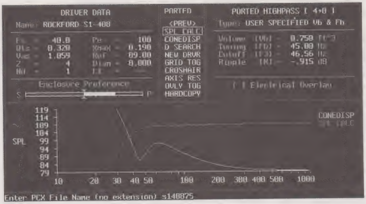




Sealed Enclosure	
Vbnet	.4 ft^3
Fc	76Hz
Q Factor	.61
Pa	100 W

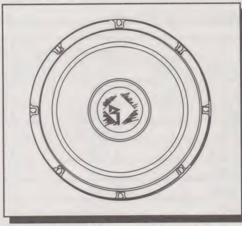


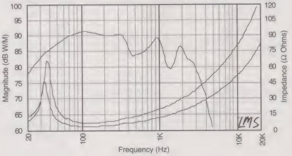
Ported Enclosure	
Vbnet	.75 ft^3
Fb	45Hz
Ripple	–.9dB
Pa	100 W
Pd	3"
Pl	10¼"



Woofer Specifications

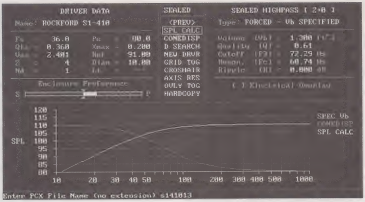
	S1-410	S1-810
Freq. Range	36Hz-2,000Hz	38Hz-2,000Hz
Impedance	4Ω	8Ω
DCR	3.4Ω	6.8Ω
FS	36Hz	38Hz
QMS	3.75	3.81
QES	.40	.42
QTS	.36	.38
SD	55.49 in.^{2} .0358 m^2	55.49 in.^{2} .0358 m^2
VAS	2.401 ft.^{3} 68 L	2.401 ft.^{3} 68 L
Voice Coil Diameter	1.5 in. 38.1 mm	1.5 in. 38.1 mm
XMAX	.20 in. 5.2 mm	.20 in. 5.2 mm
Sensitivity	91dB W/m	91dB W/m
Power Handling	100 W	100 W
VD	.06 ft.^{3} 1.6 L	.06 ft.^{3} 1.6 L





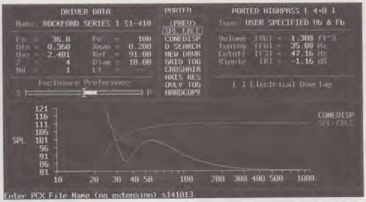
Sealed Enclosure

Vbnet	1.3 ft^3
Fc	60Hz
Q Factor	.61
Pa	80 W



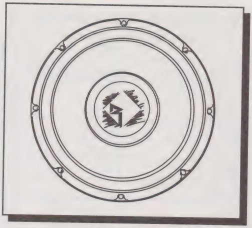
Ported Enclosure

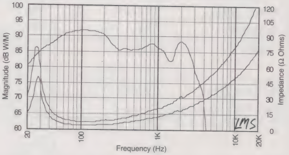
Vbnet	1.3 ft^3
Fb	35Hz
Ripple	–1.16dB
Pa	100 W
Pd	3"
Pl	9⅝"



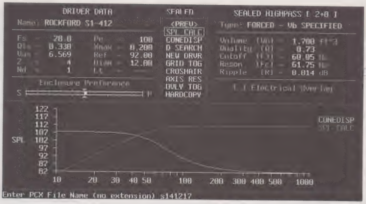
Woofer Specifications

	S1-412	S1-812
Freq. Range	28Hz-2,500Hz	29Hz-2,500Hz
Impedance	4Ω	8Ω
DCR	3.4Ω	6.7Ω
FS	28Hz	29Hz
QMS	3.41	4.11
QES	.36	.45
QTS	.33	.41
SD	84.48 in.^{2} .0545 m^2	84.48 in.^{2} .0545 m^2
VAS	6.569 ft.^{3} 186 L	6.569 ft.^{3} 186 L
Voice Coil Diameter	2 in. 50.8 mm	2 in. 50.8 mm
XMAX	.20 in. 5.1 mm	.20 in. 5.1 mm
Sensitivity	92dB W/m	92dB W/m
Power Handling	100 W	100 W
VD	.08 ft.^{3} 2.25 L	.08 ft.^{3} 2.25 L

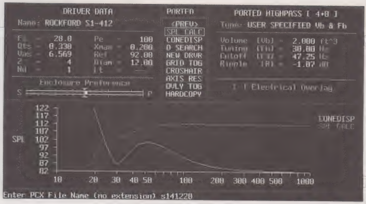




Sealed Enclosure	
Vbnet	1.7 ft^3
Fc	61Hz
Q Factor	.73
Pa	100 W

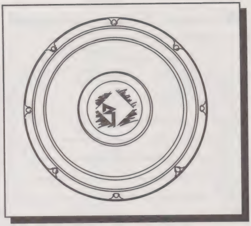


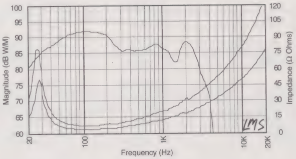
Ported Enclosure	
Vbnet	2 ft^3
Fb	30Hz
Ripple	–1.07dB
Pa	100 W
Pd	4"
Pl	15¾"



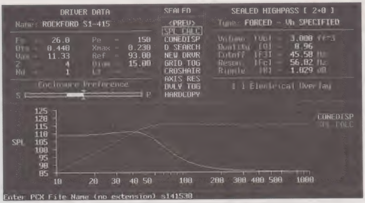
Woofer Specifications

	S1-415	S1-815
Freq. Range	26Hz-2,500Hz	26Hz-2,500Hz
Impedance	4Ω	8Ω
DCR	3.4Ω	6.7Ω
FS	26Hz	26Hz
QMS	4.89	3.90
QES	.49	.59
QTS	.44	.51
SD	134.54 in.^{2} .0868 m^2	134.54 in.^{2} .0868 m^2
VAS	11.336 ft.^{3} 321 L	11.336 ft.^{3} 321 L
Voice Coil Diameter	2 in. 50.8 mm	2 in. 50.8 mm
XMAX	.23 in. 5.9 mm	.23 in. 5.9 mm
Sensitivity	93dB W/m	93dB W/m
Power Handling	150 W	150 W
VD	.14 ft.^{3} 4 L	.14 ft.^{3} 4 L

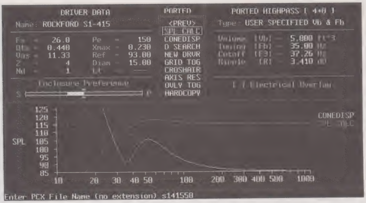




Sealed Enclosure	
Vbnet	3 ft^3
Fc	56Hz
Q Factor	.96
Pa	150 W

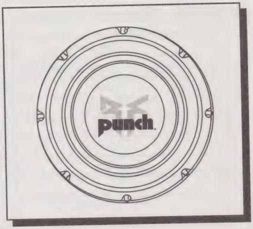


Ported Enclosure	
Vbnet	5 ft^3
Fb	35Hz
Ripple	3dB
Pa	150 W
Pd 2X	4"
Pl	6¾"

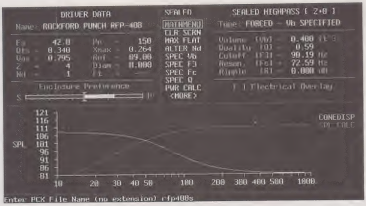


WOOFER SPECIFICATIONS

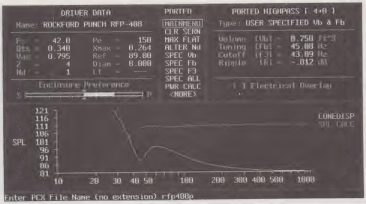
	RFP-408	RFP808
FREQ. RANGE	42Hz-1,500Hz	45Hz-1,500Hz
IMPEDANCE	4Ω	8Ω
DCR	3.4Ω	6.8Ω
FS	42Hz	45Hz
QMS	4.3	4.0
QES	.37	.37
QTS	.34	.34
SD	34.10 in.^{2} .0220 m^2	34.10 in.^{2} .0220 m^2
VAS	.795 ft.^{3} 22.5 L	.795 ft.^{3} 22.5 L
VOICE COIL DIAMETER	2 in. 50.8 mm	2 in. 50.8 mm
XMAX	.26 in. 6.7 mm	.26 in. 6.7 mm
SENSITIVITY	89dB W/M	89dB W/M
POWER HANDLING	150 W	150 W
VD	.05 ft.^{3} 1.13 L	.05 ft.^{3} 1.13 L



SEALED ENCLOSURE	
VBNET	.4 ft^3
FC	72Hz
Q FACTOR	,59
PA	150 W

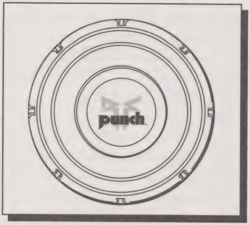


PORTED ENCLOSURE	
VBNET	.75 ft^3
FB	45Hz
RIPPLE	–.89dB
PA	150 W
PD	3"
PL	10¼"

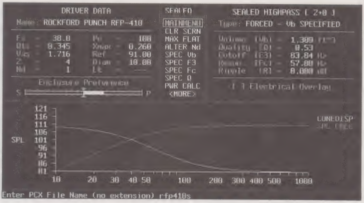


Woofer Specifications

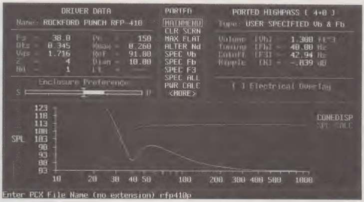
	RFP-410	RFP-810
Freq. Range	38Hz-1,500Hz	38Hz-1,500Hz
Impedance	4Ω	8Ω
DCR	3.5Ω	7Ω
FS	38Hz	40Hz
QMS	4.60	4.40
QES	.37	.42
QTS	.34	.39
SD	54.25 in.^{2} .0350 m^2	54.25 in.^{2} .0350 m^2
VAS	1.716 ft.^{3} 48.6 L	1.716 ft.^{3} 48.6 L
Voice Coil Diameter	2 in. 50.8 mm	2 in. 50.8 mm
XMAX	.26 in. 6.7 mm	.26 in. 6.7 mm
Sensitivity	91dB W/m	91dB W/m
Power Handling	150 W	150 W
VD	.06 ft.^{3} 1.7 L	.06 ft.^{3} 1.7 L



Sealed Enclosure	
Vbnet	1.3 ft^3
Fc	57Hz
Q Factor	.5
Pa	100 W

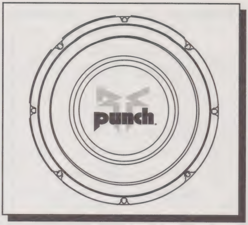


Ported Enclosure	
Vbnet	1.3 ft^3
Fb	40Hz
Ripple	–.8dB
Pa	150 W
Pd	3"
Pl	7"

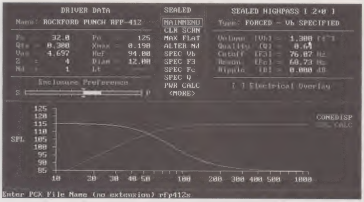


Woofer Specifications

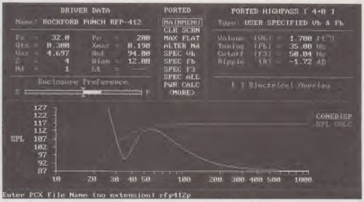
	RFP-412	RFP-812
Freq. Range	32Hz-250Hz	33Hz-250Hz
Impedance	4Ω	8Ω
DCR	3.6Ω	7.2Ω
FS	32Hz	33Hz
QMS	4.79	4.31
QES	.31	.35
QTS	.30	.32
SD	83.70 in.^{2} .0540 m^2	83.70 in.^{2} .0540 m^2
VAS	4.697 ft.^{3} 133 L	4.697 ft.^{3} 133 L
Voice Coil Diameter	2 in. 50.8 mm	2 in. 50.8 mm
XMAX	.19 in. 4.7 mm	.19 in. 4.7 mm
Sensitivity	94dB W/m	94dB W/m
Power Handling	200 W	200 W
VD	.09 ft.^{3} 2.4 L	.09 ft.^{3} 2.4 L



Sealed Enclosure	
Vbnet	1.3 ft^3
Fc	68Hz
Q Factor	.64
Pa	100 W

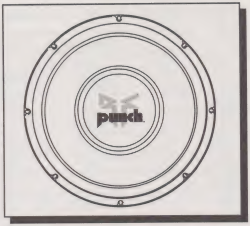


Ported Enclosure	
Vbnet	1.7 ft^3
Fb	35Hz
Ripple	−1.72dB
Pa	200 W
Pd	4"
Pl	13¾"

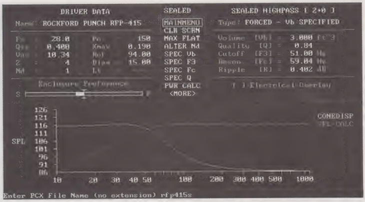


Woofer Specifications

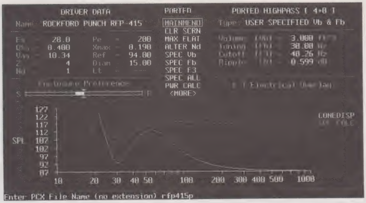
	RFP-415	RFP-815
Freq. Range	28Hz-250Hz	29Hz-250Hz
Impedance	4Ω	8Ω
DCR	3.6Ω	7.2Ω
FS	28Hz	29Hz
QMS	5.60	5.11
QES	.43	.46
QTS	.40	.42
SD	134.54 in.^{2} .0868 m^2	134.54 in.^{2} .0868 m^2
VAS	10.347 ft.^{3} 293 L	10.347 ft.^{3} 293 L
Voice Coil Diameter	2 in. 50.8 mm	2 in. 50.8 mm
XMAX	.19 in. 4.7 mm	.19 in. 4.7 mm
Sensitivity	94dB W/m	94dB W/m
Power Handling	200 W	200 W
VD	.113 ft.^{3} 3.2 L	.113 ft.^{3} 3.2 L



Sealed Enclosure	
Vbnet	3 ft^3
Fc	59Hz
Q Factor	.84
Pa	150 W

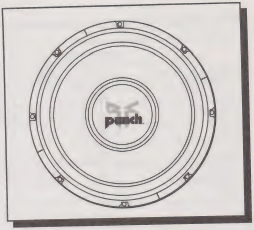


Ported Enclosure	
Vbnet	3 ft^3
Fb	30Hz
Ripple	.5dB
Pa	200 W
Pd 2X	4"
Pl	9½"

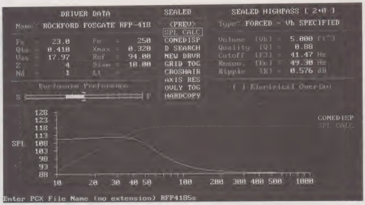


Woofer Specifications

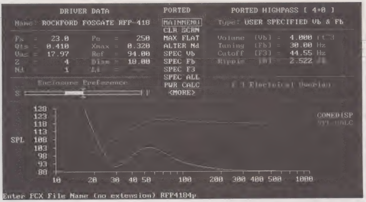
	RFP-418	RFP-818
Freq. Range	23Hz-200Hz	24Hz-200Hz
Impedance	4Ω	8Ω
DCR	3.1Ω	6.2Ω
FS	23Hz	24Hz
QMS	4.09	4.11
QES	.46	.60
QTS	.41	.53
SD	172.36 in.^{2} .1112 m^2	172.36 in.^{2} .1112 m^2
VAS	17.975 ft.^{3} 509 L	17.975 ft.^{3} 509 L
Voice Coil Diameter	2.5 in. 63.5 mm	2.5 in. 63.5 mm
XMAX	.32 in. 8.1 mm	.32 in. 8.1 mm
Sensitivity	94dB W/M	94dB W/M
Power Handling	250 W	250 W
VD	.19 ft.^{3} 5.3 L	.19 ft.^{3} 5.3 L



Sealed Enclosure	
Vbnet	5 ft^3
Fc	49Hz
Q Factor	.8
Pa	250 W

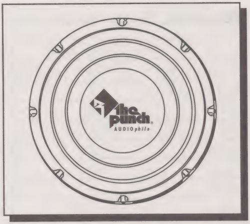


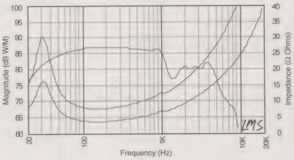
Ported Enclosure	
Vbnet	4 ft^3
Fb	30Hz
Ripple	2.5dB
Pa	250 W
Pd 2X	4"
Pl	14½"



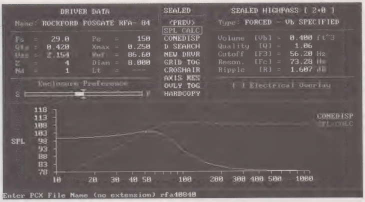
Woofer Specifications

	RFA-408	RFA-808
Freq. Range	29Hz-1,000Hz	31Hz-1,000Hz
Impedance	4Ω	8Ω
DCR	3.5Ω	7Ω
FS	29Hz	31Hz
QMS	2.08	2.19
QES	.53	.69
QTS	.42	.52
SD	34.72 in.^{2} .0224 m^2	34.72 in.^{2} .0224 m^2
VAS	2.154 ft.^{3} 61 L	2.154 ft.^{3} 61 L
Voice Coil Diameter	2 in. 50.8 mm	2 in. 50.8 mm
XMAX	.25 in. 6.3 mm	.25 in. 6.3 mm
Sensitivity	86dB W/m	86dB W/m
Power Handling	150 W	150 W
VD	.05 ft.^{3} 1.3 L	.05 ft.^{3} 1.3 L

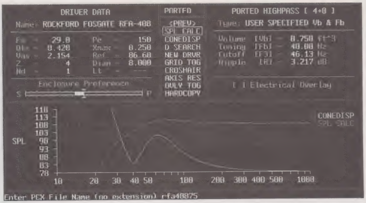




Sealed Enclosure	
Vbnet	.4 ft^3
Fc	73Hz
Q Factor	1
Pa	150 W

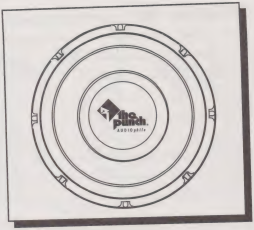


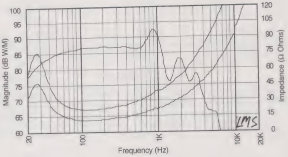
Ported Enclosure	
Vbnet	.75 ft^3
Fb	40Hz
Ripple	3dB
Pa	150 W
Pd	3"
Pl	13½"



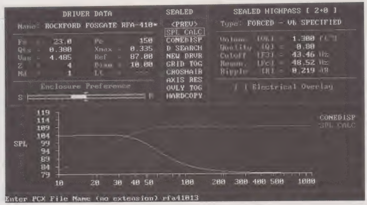
Woofer Specifications

	RFA-410	RFA-810
Freq. Range	23Hz-1,000Hz	25Hz-1,000Hz
Impedance	4Ω	8Ω
DCR	3.5Ω	7Ω
FS	23Hz	25Hz
QMS	1.94	2.03
QES	.47	.58
QTS	.38	.45
SD	54.87 in.^{2} .0354 m^2	54.87 in.^{2} .0354 m^2
VAS	4.485 ft.^{3} 127 L	4.485 ft.^{3} 127 L
Voice Coil Diameter	2 in. 50.8 mm	2 in. 50.8 mm
XMAX	.34 in. 8.5 mm	.34 in. 8.5 mm
Sensitivity	87dB W/m	86dB W/m
Power Handling	150 W	150 W
VD	.08 ft.^{3} 2.2 L	.08 ft.^{3} 2.2 L

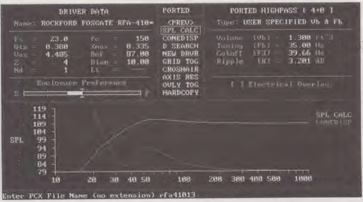




Sealed Enclosure	
Vbnet	1.3 ft^3
Fc	48Hz
Q Factor	.8
Pa	150 W

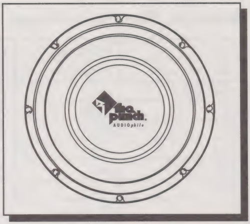


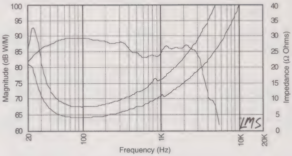
Ported Enclosure	
Vbnet	1.3 ft^3
Fb	35Hz
Ripple	3dB
Pa	150 W
Pd	3"
Pl	9⅝"



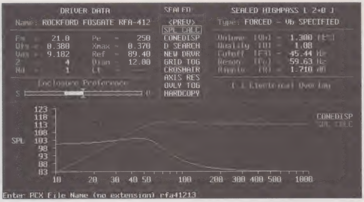
Woofer Specifications

	RFA-412	RFA-812
Freq. Range	21Hz-1,000Hz	22Hz-1,000Hz
Impedance	4Ω	8Ω
DCR	3.5Ω	7Ω
FS	21Hz	22Hz
QMS	2.20	2.80
QES	.45	.58
QTS	.38	.48
SD	84.48 in.^{2} .0545 m^2	84.48 in.^{2} .0545 m^2
VAS	9.182 ft.^{3} 260 L	9.182 ft.^{3} 260 L
Voice Coil Diameter	2.5 in. 63.5 mm	2.5 in. 63.5 mm
XMAX	.37 in. 9.3 mm	.37 in. 9.3 mm
Sensitivity	89dB W/m	89dB W/m
Power Handling	250 W	250 W
VD	.12 ft.^{3} 3.5 L	.12 ft.^{3} 3.5 L

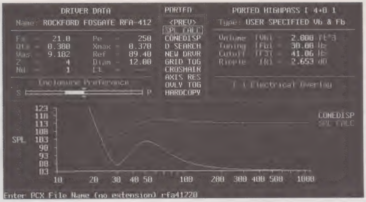




Sealed Enclosure	
Vbnet	1.3 ft^3
Fc	59Hz
Q Factor	1
Pa	250 W

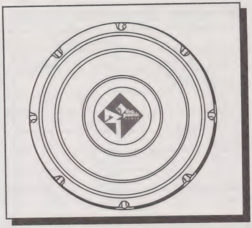


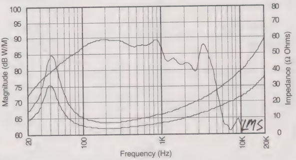
Ported Enclosure	
Vbnet	2 ft^3
Fb	30Hz
Ripple	2.6dB
Pa	250 W
Pd	4"
Pl	15¾"



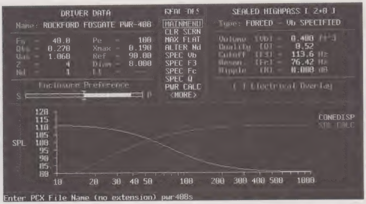
Woofer Specifications

	PWR-408	PWR-808
Freq. Range	40Hz-1,500Hz	42Hz-1,500Hz
Impedance	4Ω	8Ω
DCR	3.5Ω	7Ω
FS	40Hz	42Hz
QMS	2.44	2.35
QES	.30	.37
QTS	.27	.32
SD	32.55 in.^{2} .0210 m^2	32.55 in.^{2} .0210 m^2
VAS	1.06 ft.^{3} 30 L	1.06 ft.^{3} 30 L
Voice Coil Diameter	2 in. 50.8 mm	2 in. 50.8 mm
XMAX	.19 in. 4.8 mm	.19 in. 4.8 mm
Sensitivity	90dB W/m	90dB W/m
Power Handling	200 W	200 W
VD	.06 ft.^{3} 1.6 L	.06 ft.^{3} 1.6 L

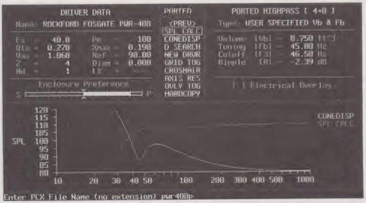




Sealed Enclosure	
Vbnet	.4 ft^3
Fc	76Hz
Q Factor	.52
Pa	100 W

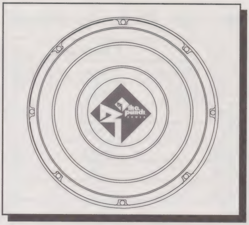


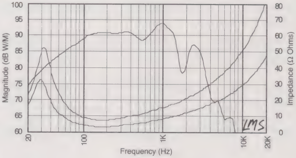
Ported Enclosure	
Vbnet	.75 ft^3
Fb	45Hz
Ripple	–2dB
Pa	100 W
Pd	2"
Pl	4"



Woofer Specifications

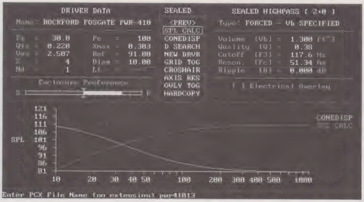
	PWR-410	PWR-810
Freq. Range	30Hz-2,500Hz	33Hz-2,500Hz
Impedance	4Ω	8Ω
DCR	3.5Ω	7Ω
FS	30Hz	33Hz
QMS	2.11	2.10
QES	.24	.32
QTS	.22	.28
SD	54.87 in.^{2} .0354 m^2	54.87 in.^{2} .0354 m^2
VAS	2.507 ft.^{3} 71 L	2.507 ft.^{3} 71 L
Voice Coil Diameter	3 in. 76 mm	3 in. 76 mm
XMAX	.30 in. 7.7 mm	.30 in. 7.7 mm
Sensitivity	91dB W/m	91dB W/m
Power Handling	300 W	300 W
VD	.13 ft.^{3} 3.6 L	.13 ft.^{3} 3.6 L





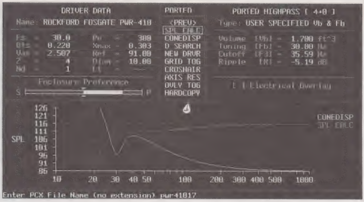
Sealed Enclosure

Vbnet	1.3 ft^3
Fc	51Hz
Q Factor	.38
Pa	100 W



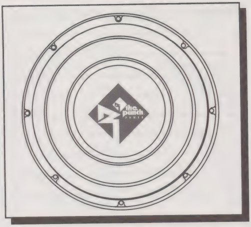
Ported Enclosure

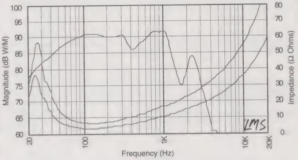
Vbnet	1.7 ft^3
Fb	30Hz
Ripple	–5.19dB
Pa	300 W
Pd	3"
Pl	10"



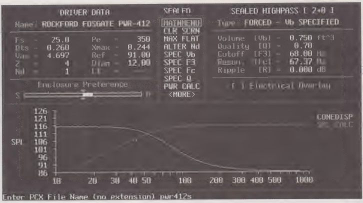
Woofer Specifications

	PWR-412	PWR-812
Freq. Range	25Hz-1,500Hz	27Hz-1,500Hz
Impedance	4Ω	8Ω
DCR	3.4Ω	6.8Ω
FS	25Hz	27Hz
QMS	2.89	2.80
QES	.28	.36
QTS	.26	.31
SD	84.48 in.^{2} .0545 m^2	84.48 in.^{2} .0545 m^2
VAS	4.697 ft.^{3} 133 L	4.697 ft.^{3} 133 L
Voice Coil Diameter	4 in. 101.6 mm	4 in. 101.6 mm
XMAX	.24 in. 6.2 mm	.24 in. 6.2 mm
Sensitivity	91dB W/m	91dB W/m
Power Handling	600 W	600 W
VD	.22 ft.^{3} 6.2 L	.22 ft.^{3} 6.2 L

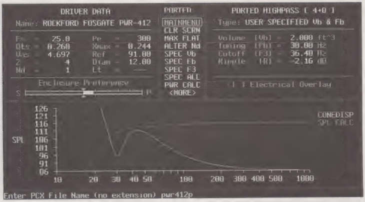




Sealed Enclosure	
Vbnet	.75 ft^3
Fc	67Hz
Q Factor	.7
Pa	350 W

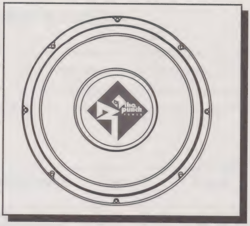


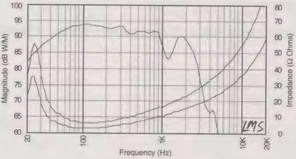
Ported Enclosure	
Vbnet	2 ft^3
Fb	30Hz
Ripple	–2dB
Pa	300 W
Pd	4"
Pl	15¾"



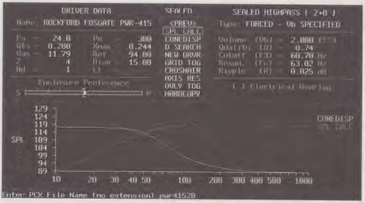
Woofer Specifications

	PWR-415	PWR-815
Freq. Range	24Hz-2,500Hz	25Hz-2,500Hz
Impedance	4Ω	8Ω
DCR	3.4Ω	6.8Ω
FS	24Hz	24Hz
QMS	3.20	2.86
QES	.31	.34
QTS	.28	.31
SD	134.54 $in.^2$ .0868 m^2	134.54 $in.^2$ .0868 m^2
VAS	11.795 $ft.^3$ 334 L	11.795 $ft.^3$ 334 L
Voice Coil Diameter	4 in. 101.6 mm	4 in. 101.6 mm
XMAX	.24 in. 6.2 mm	.24 in. 6.2 mm
Sensitivity	94dB W/m	94dB W/m
Power Handling	600 W	600 W
VD	.26 $ft.^3$ 7.3 L	.26 $ft.^3$ 7.3 L

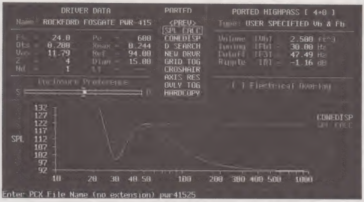




Sealed Enclosure	
Vbnet	2 ft^3
Fc	63Hz
Q Factor	.74
Pa	300 W



Ported Enclosure	
Vbnet	2.5 ft^3
Fb	30Hz
Ripple	−1.16dB
Pa	600 W
Pd	4"
Pl	12"



Woofer Quick Reference

Low Frequency Speakers: Subwoofers are available at 4 or 8 Ohms impedance. The enclosure volume is given in cubic feet, the tuning frequency is in Hertz. Woofers come packaged with the speaker and manual. Power rating is given for continuous Watts RMS. The metric sizes are given in liters for the volume listing and millimeters for Mounting Diameter and Mounting Depth.

Series 1

Model	Box	Volume Min-Max	Tuning	Power	Mounting Diameter	Mounting Depth
"NEW" S1-408/808	P	0.5-1.0 *(14-28L)*	45Hz	100	$7\frac{1}{8}$ *(191)*	$3\frac{11}{16}$ *(94)*
"NEW" S1-410/810	P	0.75-1.5 *(22-44L)*	40Hz	100	$9\frac{1}{4}$ *(235)*	$3\frac{9}{32}$ *(99)*
"NEW" S1-412/812	P	1.25-2.0 *(35-70L)*	35Hz	100	11 *(279)*	$4\frac{13}{16}$ *(122)*
"NEW" S1-415/815	S/P	2.5-4.0 *(71-113L)*	30Hz	150	14 *(355.6)*	$6\frac{3}{32}$ *(155)*

Rockford Fosgate Punch

Model	Box	Volume Min-Max	Tuning	Power	Mounting Diameter	Mounting Depth
"NEW" RFP-408/808	P	0.5-1.0 *(14-28L)*	45Hz	150	$6\frac{31}{32}$ *(177)*	$3\frac{23}{32}$ *(95)*
"NEW" RFP-410/810	P	0.75-1.5 *(22-44L)*	40Hz	150	$9\frac{1}{4}$ *(235)*	$4\frac{1}{2}$ *(114)*
"NEW" RFP-412/812	P	1.25-2.0 *(35-70L)*	35Hz	200	11 *(279)*	$5\frac{1}{8}$ *(130)*
"NEW" RFP-415/815	S/P	2.5-4.0 *(71-113L)*	30Hz	200	14 *(355.6)*	$6\frac{1}{2}$ *(165)*
"NEW" RFP-418/818	S/P	4.0-6.0 (113-170L)	30Hz	250	$16\frac{21}{32}$ *(423)*	$7\frac{3}{4}$ *(197)*

Audiophile

Model	Box	Volume Min-Max	Tuning	Power	Mounting Diameter	Mounting Depth
"NEW" RFA-408/808	S	0.33-.75 *(9-21L)*	45Hz	150	$7\frac{3}{8}$ *(187)*	$3\frac{5}{8}$ *(92)*
"NEW" RFA-410/810	S	0.66-1.0 *(19-28L)*	35Hz	150	$9\frac{11}{32}$ *(237)*	$4\frac{13}{16}$ *(122)*
"NEW" RFA-412/812	S	0.88-1.25 *(25-35L)*	30Hz	250	11 *(279)*	$5\frac{1}{16}$ *(129)*

Power

Model	Box	Volume Min-Max	Tuning	Power	Mounting Diameter	Mounting Depth
PWR-408/808	P	1.0 *(28L)*	45Hz	200	$6\frac{31}{32}$ *(177)*	$4\frac{3}{16}$ *(106)*
PWR-410/810	P	1.0-1.5 *(28-44L)*	40Hz	300	$9\frac{19}{32}$ *(239)*	$4\frac{7}{8}$ *(124)*
PWR--412/812	P	1.5-2.0 *(44-70L)*	35Hz	600	$10\frac{15}{16}$ *(278)*	$5\frac{9}{32}$ *(142)*
PWR-415/815	P	2.5-3.0 *(71-85L)*	30Hz	600	$13\frac{29}{32}$ *(353)*	$6\frac{25}{32}$ *(172)*

BOX THAT ROCKS!

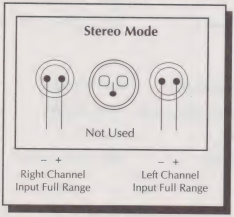
BTR-9010

(2) 12" WOOFERS

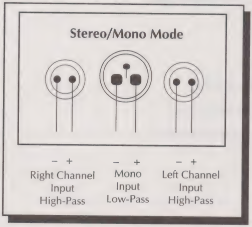
(2) 5" COAXIAL

POWER HANDLING:	350 WATTS
FREQUENCY RESPONSE:	58HZ-20KHZ
SENSITIVITY:	95DB
DIMENSIONS:	32" x 17¼" x 18"

STEREO WIRING CONFIGURATION



STEREO/MONO WIRING CONFIGURATION



BOX THAT ROCKS!

BTR-2152

(2) 15" Woofers

(2) Horn Tweeters

Power Handling:	350 Watts
Frequency Response:	53Hz-20kHz
Sensitivity:	96dB
Dimensions:	32" x 19½" x 20½"

BTR-2122

(2) 12" Woofers

(2) Horn Tweeters

Power Handling:	350 Watts
Frequency Response:	50Hz-20kHz
Sensitivity:	96dB
Dimensions:	32" x 17¼" x 18"

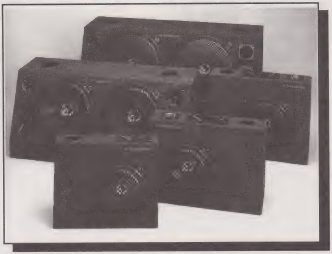
BTR-2102

(2) 10" Woofers

(2) Horn Tweeters

Power Handling:	200 Watts
Frequency Response:	48Hz-20kHz
Sensitivity:	94dB
Dimensions:	32" x 14" x 14½"

BOX THAT ROCKS!



BTR-153		Power Handling:	350 Watts
(2) 15" Woofers		Frequency Response:	45Hz-20kHz
(2) Horn Tweeters		Sensitivity:	93dB
		Dimensions:	42" x 19½" x 16¼"
BTR-123		Power Handling:	350 Watts
(2) 12" Woofers		Frequency Response:	45Hz-20kHz
(2) Midrange Horns		Sensitivity:	93dB
(2) Horn Tweeters		Dimensions:	36" x 16¾" x 14"
BTR-103		Power Handling:	200 Watts
(2) 10" Woofers		Frequency Response:	43Hz-20kHz
(2) Midrange Horns		Sensitivity:	93dB
(2) Horn Tweeters		Dimensions:	20" x 15½" x 7½"
BTR-83		Power Handling:	200 Watts
(2) 8" Woofers		Frequency Response:	40Hz-20kHz
(2) Midrange Horns		Sensitivity:	93dB
(2) Horn Tweeters		Dimensions:	20" x 14" x 7¼"
BTR-82		Power Handling:	200 Watts
(2) 8" Woofers		Frequency Response:	43Hz-20kHz
(2) Horn Tweeters		Sensitivity:	93dB
		Dimensions:	17" x 13" x 7¼"

BOX THAT ROCKS!

BTR-416

(1) 6½" WOOFER

POWER HANDLING: 100 WATTS
FREQUENCY RESPONSE: 48Hz-150Hz
SENSITIVITY: 89dB
DIMENSIONS: 7" x 19¼"
INCLUDES MOUNTING STRAP

BTR-418

(1) 8" WOOFER

POWER HANDLING: 100 WATTS
FREQUENCY RESPONSE: 40Hz-150Hz
SENSITIVITY: 89dB
DIMENSIONS: 8¾" x 21"
INCLUDES MOUNTING STRAP

BTR-4110

(1) 10" WOOFER

POWER HANDLING: 100 WATTS
FREQUENCY RESPONSE: 32Hz-150Hz
SENSITIVITY: 91dB
DIMENSIONS: 10¾" x 23"
INCLUDES MOUNTING STRAP

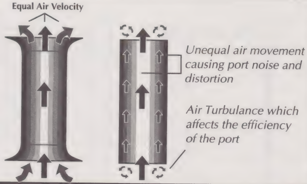
AEROPORT TUBE

3" AND 4" DIAMETERS

12" TUBE LENGTH

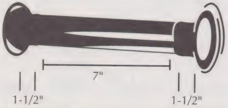
3 PIECE ASSEMBLY

Impedance matches the air inside the enclosure with the port to the air outside increasing efficiency which increases power handling and SPL with less distortion enabling a 10% air velocity which is a figure given in Term Pro.



CALCULATING PORT LENGTH

Example: 10" Total Usable Port Length

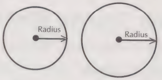


After the enclosure has been designed, the following formula may be useful when calculating the port length:

$$Lv = \frac{8466.4R^2}{(Vb)Fb^2} - 1.463R$$

Lv = port length in inches
Vb = volume of box in cubic feet
Fb = tuning frequency
R = radius of port

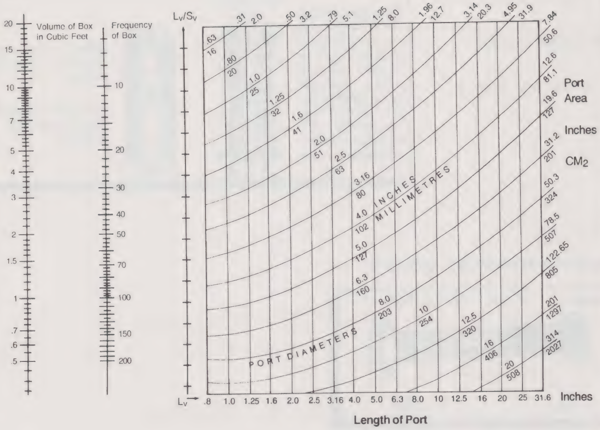
PCH-AP3	PCH-AP4



R=1.34"	R=2.03"
D=2.68"	D=4.06"

When cutting the Aeroport center tube, the length should be cut 3" shorter than calculated.

NOMOGRAM



USING THE PORT TUNING NOMOGRAM

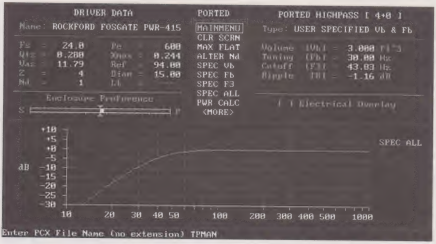
Use this chart by drawing a line connecting the box volume and the desired port frequency, and extending it through the third vertical column, labeled Lv/Sv. From the third vertical column, draw a horizontal line across the page. The curved lines that your horizontal line intersect indicate the diameter of ports that can be used (in inches and centimeters). Areas are given at the margin for use with rectangular ports. To get the port length, draw a vertical line from the place your horizontal line intersects the curved line for the diameter, to the bottom of the chart where the length shown. Add 10% to the port length the chart gives. Boxes always act smaller than third measured volume would indicate.

TERM-PRO

COMPUTER-AIDED ENCLOSURE SOFTWARE PROGRAM

TERM-PRO is a computer-aided enclosure design software program. The function of the program is to assist you in the development of speaker enclosures. The program will model Driver/Enclosures relationships with many parameters, i.e. Box Volume, Resonant Frequency, Tuning Frequency, 3dB down point and many others.

The primary use of **TERM-PRO** involves using either a three button mouse or the arrow keys on the keyboard. Very little typing is necessary to use **TERM-PRO**. Once the program is open and running, you will see a box highlighting a particular menu selection. Pressing ***Enter*** will activate that function and take you to that sub menu if one applies. The first menu is the MAINMENU which allows you to choose the direction you may want to take with your box building methods. The most common way is to first select a driver. To do that, use the mouse or the arrow keys to move the highlight box down to DATA BASE and press ***Enter***. All menu selections will be made using this manner unless otherwise noted.



DATABASE The data base is comprised of approximately 900 drivers and is labeled EXPANDED. From here you can break out those drivers into 21 different libraries which will hold 1000 drivers each, equaling 21,000 drivers in all.

SEARCH & SCROLL Allows you to sort quickly through the data base.

HARDCOPY Allows you to print what is on the screen.

USING & UNDERSTANDING THE "F-KEYS" Gives you a faster and easier way to be more effective using **TERM-PRO**.

Here is a detailed look at a few of the more special or unique menu items:

ENCL DES Once you have selected ENCL DES the screen will be filled with a pictorial display of the different enclosures, 20 designs in all. Use the page down button to see the second page.

MAX FLAT Calculates a curve with a Q of .707 which is a Butterworth alignment.

PWR CALC Shows the relative power the speaker can handle in the enclosure you design.

SPL CALC Remember: the curves are based on mathematical calculations of free field measurements with power applied to the speaker in that enclosure you create.

CONEDISP Shows a graphic display of the cone movement a driver will have with power applied.

NEW DRVR Allows you to select a new driver without returning to the DATABASE menu.

PORT DES Develops the porting information needed for proper tuning of your enclosures.

CHAMBER When designing a BANDPASS enclosure there may be a port on both sides of the enclosure. Use this function to switch from one side to the other.

FIND FB If you know the box volume in Ft3 and the port dimensions you can find the tuning frequency of any box.

WOOD DES Illustrates the construction of the enclosure in board layout dimensions and 3D mode (MODEL 3D).

CALV VOL If you know all measurements of the enclosure (square, rectangle, wedge 1 or wedge 2 & bandpass) the program will tell you the volume of the box in cubic feet.

XOVER DES Offers HIGHPASS, LOWPASS and BANDPASS types of crossovers. BUTTERWORTH alignments have 6dB, 12dB & 18dB slopes and the CHEBYCHEV BEC BESSEL & LINKWITZ-RILEY are all available in the 12dB slopes. Series Notch and Zobel are also available.

STANDARD Converts the pure values of components to a value readily available at your local electronics store and also plots the difference. Many time the curves are not all that different.

OVLY DES Serves as a storage bank for acoustical responses of automobiles or different crossover designs such as subsonic filters that can be overlaid into enclosure response. This will save as many as 100 different plots.

CHAPTER 5
CROSSOVERS

- CROSSOVER DESIGNS
- ZOBEL NETWORK
- SERIES NOTCH FILTER
- OPTICAL COMPRESSION
- PUNCH PASSIVE CROSSOVERS
- PUNCH ACTIVE CROSSOVERS
- SYMMETRY EPX_2
- BUTTERWORTH ALIGNMENT CHART

Crossovers

Crossover Designs

- Active Crossovers
- Passive Crossovers

In multiple speaker systems, speakers have different operating bandwidths and frequency response. A crossover filters the signal which assists in the control of the speaker's response. A crossover is either **Active** or **Passive**. Crossovers are defined by **Filter Resonance**, **Slope** and **Q**.

Active crossovers require power to operate and filter the signal before amplification. Active crossovers offer the ability of bi-amplification and tri-amplification in a system to divide the power to different speakers.

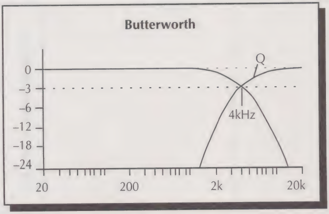
Passive crossovers are made up of L/C (inductance and capacitance) that filter the signal after amplification. These crossovers are dependent on the impedance of the speaker to be stable in order to filter properly. Passive crossovers enable the system to be powered by one amplifier.

Filter Resonance of a crossover is the frequency at which the component reactances are equal and designate the crossover point.

Slope is the amount of attenuation per octave from the crossover point. These attenuations are also referred by order of slope. 1st order: 6dB, 2nd order: 12dB, 3rd order: 18dB, 4th order: 24dB.

"Q" describes the shape of the "knee" of the roll-off response. The Q can be in different shapes, such as Chebychev .1, Butterworth .707, Bessel .58, Linkwitz-Riley .49. The crossover Q can assist in the placement of the speakers in the vehicle.

In a two-way passive system with a Butterworth alignment. The signal is divided into a low and high frequency output. At the intersection of the low and high-pass filter there is a 3dB reduction. The two filters will combine for an increase of 3dB at the crossover point. This will sum a flat response in the crossover region.

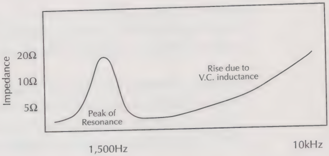


Passive Crossovers

Zobel Network

- Maximizes the control of the crossover network

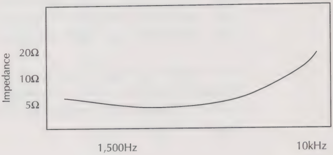
This circuit is designed to flatten the impedance response rise caused by the voice coil inductance. This will improve the low-pass crossover response and reduce the harshness of the tweeter. The use of this circuit enables it to have a stable platform for the use of an L-Pad.



Series Notch Filter

- Flatten response of the speaker

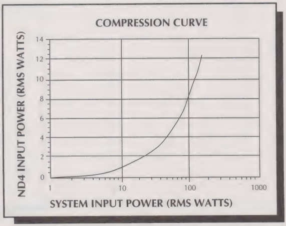
This circuit will dampen and eliminate the effect that driver's resonance can cause on the crossover response.



Optical Compression

- Protects tweeter from increased power

The circuit applies soft limiting to the input to the tweeter. If the power applied reaches the threshold of the tweeter 14 volts rms (50 watts) the circuit will begin to compress up to 20 volts rms (100 watts). This circuit will also illuminate to absorb any DC current present from a clipped output from an amplifier.



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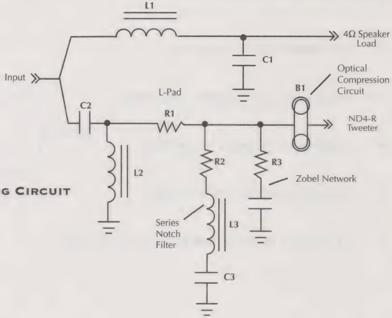
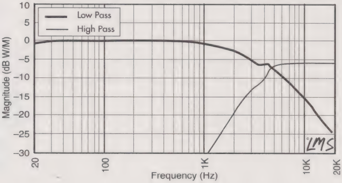
Passive Crossovers

2X-4

Features

- Linkwitz-Riley Alignment
- 12dB/octave at 4kHz
- Series Notch Filter
- Zobel Impedance Compensating Circuit
- Optical Compression Circuit
- High-Pass –6dB L-Pad
- 12 Gauge Terminal Connectors
- Mylar Capacitors
- Iron Core Coils
- Packaged with RFA-414, 514, 614

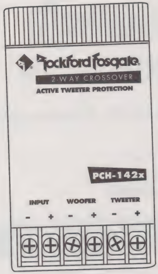




punch

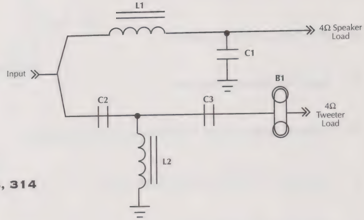
Passive Crossovers

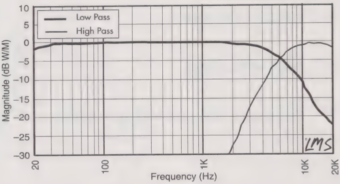
PCH-142X



Features

- Butterworth Alignment
- 12dB/octave at 6kHz Low-Pass
- 18dB/octave 6kHz High-Pass
- Active Tweeter Protection
- Packaged with PCH-614, 514, 414, 314

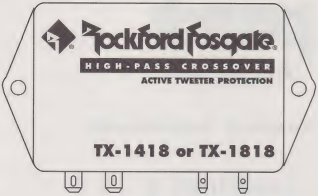




punch

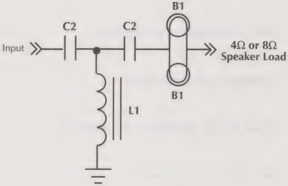
Passive Crossovers

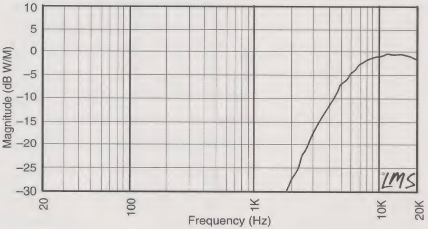
TX-1418/1818



FEATURES

- Butterworth Alignment
- 18dB/octave at 6kHz
- Polarized Faston Terminals
- Mylar Capacitors
- Active Tweeter Protection





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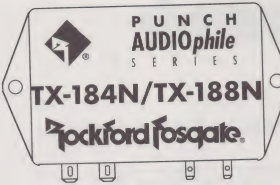
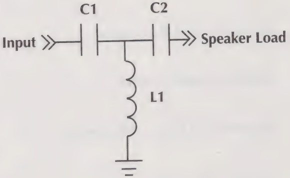
Passive Crossovers

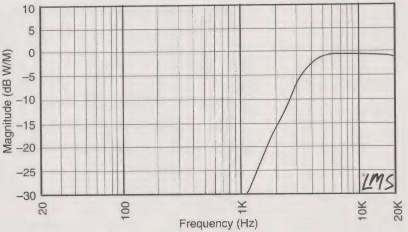


184N/188N

Features

- Butterworth Alignment
- 18dB/octave at 4kHz
- Polarized Faston Terminals
- Mylar Capacitors
- Included wtih ND-4 Tweeter





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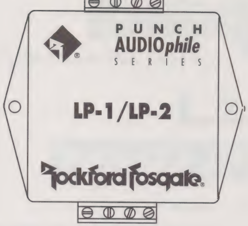
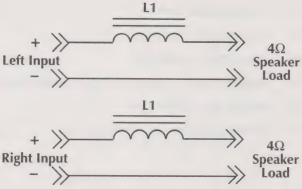
PASSIVE CROSSOVERS

LP-1/LP-2

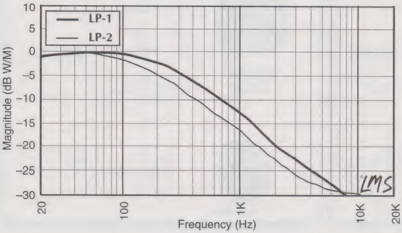


FEATURES

- BUTTERWORTH ALIGNMENT
- 6dB/OCTAVE
- TERMINAL BLOCK CONNECTOR
- (2) IRON CORE COILS



	STEREO			MONO		
	2Ω	4Ω	8Ω	2Ω	4Ω	8Ω
LP-1	100Hz	200Hz	400Hz	50Hz	100Hz	200Hz
LP-2	50Hz	100Hz	200Hz	25Hz	50Hz	100Hz

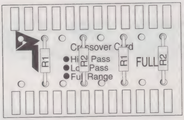


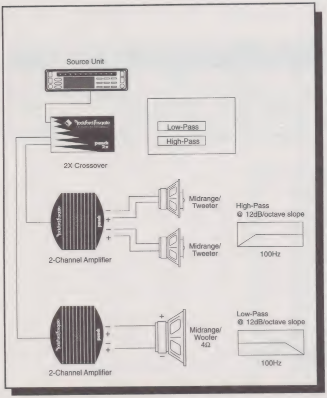
punch
Active Crossovers

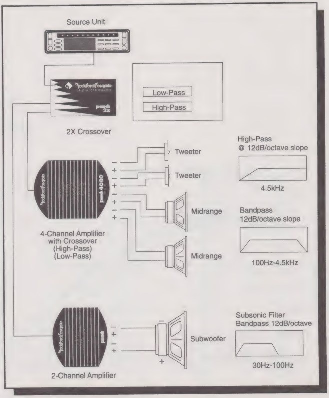
2X

Features

- 3 Year Warranty
- 12dB/Octave Slope Butterworth Alignment
- 2-Way Selectable Crossover
- 2 Channel Input
- 4 Channel Output
- Gold-Plated RCAs
- Individually Selectable High-Pass, Low-Pass and Full Range
- Unity Gain Output
- Butterworth Alignment





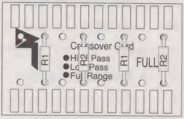


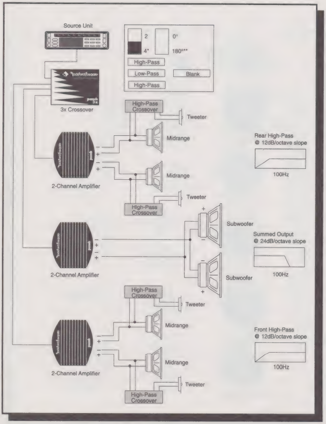
punch
Active Crossovers

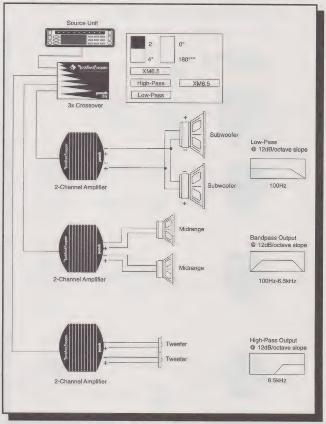
3X

Features

- 3 Year Warranty
- 12dB/Octave Slope Butterworth Alignment
- 3-Way Selectable Crossover
- 4 Channel Input
- 6 Channel Output
- Gold-Plated RCAs
- (2) Individually Selectable High-Pass, Low-Pass and Full Range
- (1) Dual Filtred Non-Faded Output with Phase Reversal Switch
- Unity Gain Output
- Butterworth Alignment





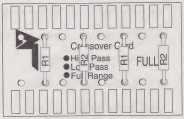


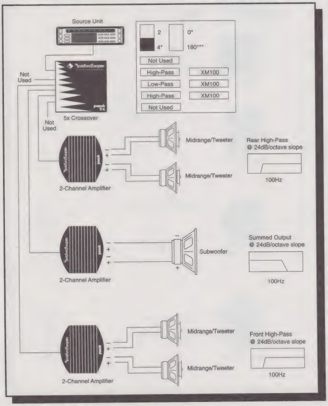
punch
ACTIVE CROSSOVERS

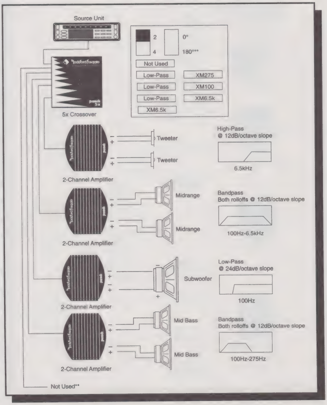
5X

FEATURES

- 3 YEAR WARRANTY
- 12dB/OCTAVE SLOPE BUTTERWORTH ALIGNMENT
- 5-WAY SELECTABLE CROSSOVER
- 4 CHANNEL INPUT
- 10 CHANNEL OUTPUT
- GOLD-PLATED RCAS
- (2) INDIVIDUALLY SELECTABLE HIGH-PASS, LOW-PASS AND FULL RANGE
- (3) DUAL FILTRED NON-FADED OUTPUT WITH PHASE REVERSAL SWITCH
- SUMMED OUTPUT PHASE REVERSAL SWITCH
- UNITY GAIN OUTPUT
- BUTTERWORTH ALIGNMENT





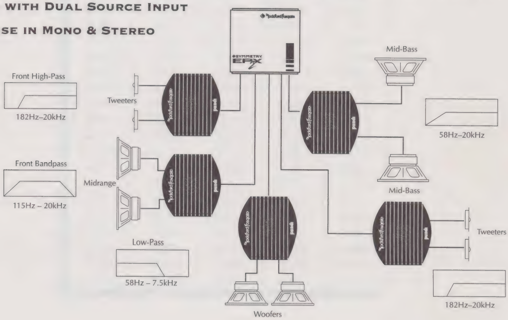


SYMMETRY EPX2

EPX2

FEATURES

- 4 GLOBAL PRESETS FOR: EQUALIZER, CROSSOVER, PRE-AMP
- VDISC "VOLUME DEPENDENT INTERACTIVE SYSTEM"
- ACTIVATES A FOUR VOLUME PRESET
- ILLUMINATED RDAT
- 5-WAY 10 CHANNEL CROSSOVER
- E14 BAND STEREO EQUALIZER
  - 16 PRESETS WITH CHARACHTERS
  - HALF OCTAVE CENTERS: 30, 45, 60, 90, 125, 180, 250, 375, 500HZ
  - OCTAVE CENTERS: 1K, 2K, 4K, 8K, 16KHZ
- E28 BAND STEREO EQUALIZER
  - 20 TOTAL PRESETS (16 NORMAL, 4 VDISC)
  - 1/3 OCTAVE CENTERS: 32, 40, 50, 63, 80, 100, 125, 160, 200, 250, 320, 400, 500, 630, 800HZ, 1K, 1.25K, 1.6K, 2K, 2.5K, 3.25K, 4K, 5K, 6.3K, 8K, 10K, 12.5K, 16K
- PRE-AMP WITH DUAL SOURCE INPUT
- PINK NOISE IN MONO & STEREO



Punch Active Crossovers

- The Crossover Module programmed with a Butterworth Alignment can be used High-Pass or Low-Pass without changing any resistor values.

- This chart is to be used with the **XM00** card or with a card that has a capacitor value of .022µF.

Butterworth Alignment

Use 5% resistors in conjunction with .022µF standard capacitor.

	Low Pass		High Pass	
Freq.	R1	R2	R1	R2
18.5Hz	390k Ω	390k Ω	390k Ω	390k Ω
26Hz	270k Ω	270k Ω	270k Ω	270k Ω
33Hz	220k Ω	220k Ω	220k Ω	220k Ω
40Hz	180k Ω	180k Ω	180k Ω	180k Ω
48Hz	150k Ω	150k Ω	150k Ω	150k Ω
60Hz	120k Ω	120k Ω	120k Ω	120k Ω
72Hz	100k Ω	100k Ω	100k Ω	100k Ω
88Hz	82k Ω	82k Ω	82k Ω	82k Ω
106Hz	68k Ω	68k Ω	68k Ω	68k Ω
130Hz	56k Ω	56k Ω	56k Ω	56k Ω
154Hz	47k Ω	47k Ω	47k Ω	47k Ω
185Hz	39k Ω	39k Ω	39k Ω	39k Ω
220Hz	33k Ω	33k Ω	33k Ω	33k Ω
270Hz	27k Ω	27k Ω	27k Ω	27k Ω
330Hz	22k Ω	22k Ω	22k Ω	22k Ω
400Hz	18k Ω	18k Ω	18k Ω	18k Ω
480Hz	15k Ω	15k Ω	15k Ω	15k Ω
600Hz	12k Ω	12k Ω	12k Ω	12k Ω
720Hz	10k Ω	10k Ω	10k Ω	10k Ω
880Hz	8.2k Ω	8.2k Ω	8.2k Ω	8.2k Ω
1.06kHz	6.8k Ω	6.8k Ω	6.8k Ω	6.8k Ω
1.3kHz	5.6k Ω	5.6k Ω	5.6k Ω	5.6k Ω
1.54kHz	4.7k Ω	4.7k Ω	4.7k Ω	4.7k Ω
1.85kHz	3.9k Ω	3.9k Ω	3.9k Ω	3.9k Ω
2.2kHz	3.3k Ω	3.3k Ω	3.3k Ω	3.3k Ω
2.7kHz	2.7k Ω	2.7k Ω	2.7k Ω	2.7k Ω
3.3kHz	2.2k Ω	2.2k Ω	2.2k Ω	2.2k Ω
4.0kHz	1.8k Ω	1.8k Ω	1.8k Ω	1.8k Ω
4.8kHz	1.5k Ω	1.5k Ω	1.5k Ω	1.5k Ω
6.0kHz	1.2k Ω	1.2k Ω	1.2k Ω	1.2k Ω
7.2kHz	1.0k Ω	1.0k Ω	1.0k Ω	1.0k Ω
8.8kHz	820 Ω	820 Ω	820 Ω	820 Ω
10.6kHz	680 Ω	680 Ω	680 Ω	680 Ω
13.0kHz	560 Ω	560 Ω	560 Ω	560 Ω
15.4kHz	470 Ω	470 Ω	470 Ω	470 Ω

Chapter 6
Understanding Crossovers

- Crossover Filters
- Crossover Terms
- Crossover Orders
- Crossover "Q"

UNDERSTANDING CROSSOVERS

CROSSOVER FILTERS

- **WHY CROSSOVERS?**

Ideally, the best sounding system would have a single speaker capable of flat frequency response and power handling at all frequencies in the audio bandwidth. Unfortunately, no single speaker exists that can perform this task. The solution is to use several speakers to play different bandwidths in the audio frequencies. To accomplish this, crossovers are used. A crossover is a filter that modifies the audio signal as a function of frequency. Crossovers divide the audio signal sending separate bandwidths of audio information. In a passive system, the crossover comes after the amplifier but before the speaker. In an active crossover, the crossover comes after the source unit but before the amplifiers. *Figure 6.1* displays both active and passive crossovers as they would appear in a signal chain.

FLOW CHARTS FOR PASSIVE AND ACTIVE CROSSOVERS

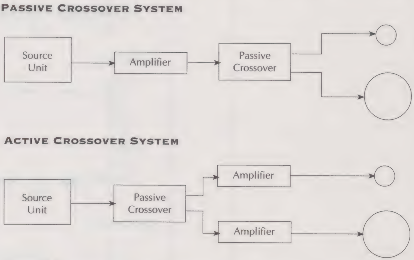


Figure 6.1

CROSSOVER TYPES

- HIGH-PASS
- LOW-PASS
- BANDPASS

There are three types of crossover operations: **high-pass, low-pass** and **bandpass** *(Figure 6.2)*. In a **high-pass** crossover, high frequency information passes through the circuit and low frequency information is filtered off as the frequency decreases. A **low-pass** crossover passes low frequency information and attenuates the output as the frequency increases. It works just the opposite of a high-pass filter. A **bandpass** crossover is a combination of a high-pass filter and low-pass filter. Crossovers can be *Digital, Active, Passive* and *Acoustical* in design. To better understand crossovers, let's examine the terms used to define them.

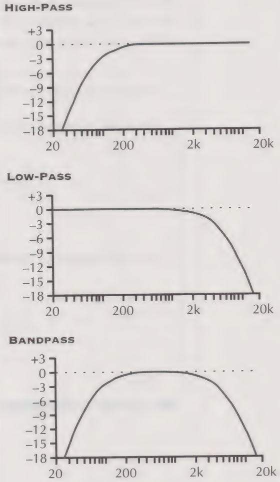


Figure 6.2

Crossover Terms

- **Decibels or dB**

One way we measure sound is its perceived loudness. The measurement standard we use is the *decibel* or *dB*. Decibels are measured on a relative loudness logarithmic scale. A logarithmic scale differs from a linear scale in its rate of change. See *Table 7.1* for a relationship between dB power applied and the difference in perceived loudness.

dB Output	90dB	100dB	103dB	110dB
Power Applied	10 Watts	100 Watts	200 Watts	1000 Watts
Difference in Perceived Loudness	Half as loud requiring only one tenth the power to achieve this output level	Reference	Noticeable difference in volume requiring twice the power	Sounds twice as loud but requires ten times the power

Table 7.1

- **Octave**

An **octave** is the halving or doubling of a given frequency. Octave intervals are perceived as equal pitch intervals. If 100Hz is the starting frequency, f_o, then half or twice the starting frequency is considered to be one octave away. *(Figure 6.3)*

50Hz	← one octave →	100Hz	← one octave →	200Hz
$1/2\ f_o$		f_o		$2\ f_o$

Figure 6.3

- **Slope Rate (dB/Octave)**

Slope rate (dB/octave) is the rate at which the crossover filters the output level relative to the frequency in the audio signal. Example: The output of the speaker is 100dB in the pass band. A high-pass crossover at 100Hz at 12dB/octave is used. *Figure 6.4* displays output level versus frequency.

Output Level versus Frequency...

50Hz		100Hz		200Hz
85dB	←→	97dB	←→	100dB

Figure 6.4

• Crossover Point or F_3

The **Crossover Point (F_3)** is the point in a Butterworth Crossover Alignment where the acoustical output of the crossover is –3dB from the reference level. *Figure 6.5* is a three-way system configuration with Butterworth alignments crossover points at 100Hz and 4,000Hz.

CROSSOVER POINTS AS 100HZ AND 4,000HZ

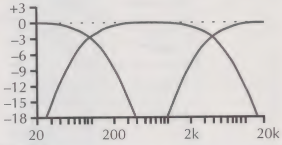


Figure 6.5

Crossover Orders

The **Crossover Order** refers to the number of basic components in a passive crossover filter. The crossover order also states the slope rate of the crossover filter in dB/octave. For different types of Crossover Orders, refer to *Table 6.2, Figure 6.6* and *Figure 6.7*.

CROSSOVER ORDER

n x 6dB/octave where **n** is the order of the crossover

Examples:

1st order	=	1 x 6dB/octave =>	6dB/octave
2nd order	=	2 x 6dB/octave =>	12dB/octave
3rd order	=	3 x 6dB/octave =>	18dB/octave

Table 6.2

1ST, 2ND AND 3RD ORDER CROSSOVER SLOPES

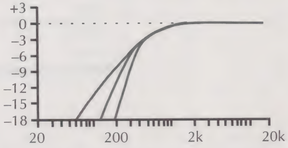


Figure 6.6

• **1ST THROUGH 4TH ORDER PASSIVE CROSSOVERS**

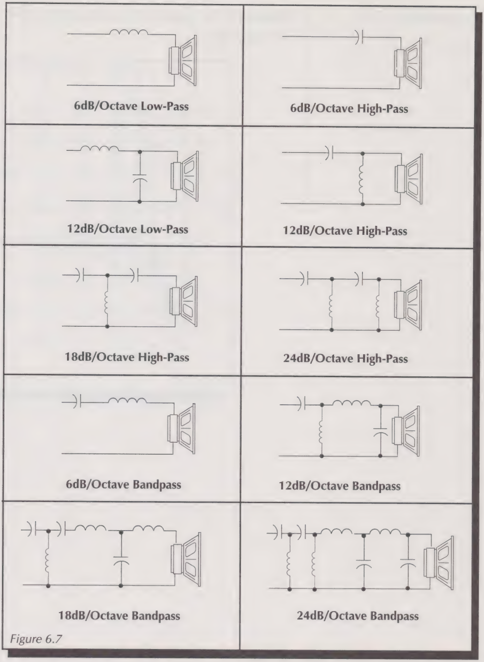


Figure 6.7

CROSSOVER "Q"

Crossover "Q" refers to the shape of the roll-off at the crossover frequency. See *Figure 6.8*.

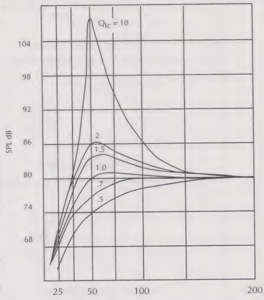


Figure 6.8

CHAPTER 7

PASSIVE COMPONENT SYSTEMS

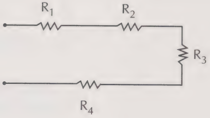
- OHM'S LAWS
- VOLTAGE & CURRENT
- FORMULAS
- PASSIVE CROSSOVER REFERENCE CHARTS
- IMPEDANCE COMPENSATION NETWORKS
- LEVEL ATTENUATION NETWORKS
- FREQUENCY SHAPING NETWORKS
- CROSSOVER DESIGN

Passive Component Systems

Ohms Law

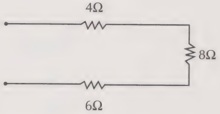
• Series Circuits

$$R_T = R_1 + R_2 + R_3 \ldots$$



Example:

$$R_T = 4\Omega + 8\Omega + 6\Omega = 18\Omega$$



Note: R_T is always larger than the largest resistance in the circuit, i.e. $R_T = 18\Omega > 8\Omega$

• Parallel Circuits

$$R_T = \cfrac{1}{\cfrac{1}{R_1} + \cfrac{1}{R_2} + \cfrac{1}{R_3} \ldots}$$

Example:

$$R_T = \cfrac{1}{\cfrac{1}{8} + \cfrac{1}{8} + \cfrac{1}{4}} \Rightarrow \cfrac{1}{\cfrac{2}{8} + \cfrac{1}{4}} \Rightarrow$$

$$\cfrac{1}{\cfrac{1}{4} + \cfrac{1}{4}} \Rightarrow \cfrac{1}{\cfrac{2}{4}} \Rightarrow 2\Omega$$

Note: R_T is always smaller than the smallest resistance in the circuit, i.e. $R_T = 2\Omega < 4\Omega$

For two resistors: $R_T = \dfrac{R_1 \times R_2}{R_1 + R_2}$

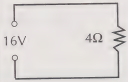
$V = IR$

Voltage = Current x Resistance

$R = \frac{V}{I}$ and $I = \frac{V}{R}$

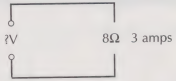
Note: These laws apply to both AC (alternating current) and DC (direct current) circuits.

Example:



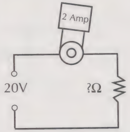
$V_T = 16V$
$R_T = 4\Omega$
$I_T = ?$

$I = \frac{V}{R}$ or $\frac{16}{4} = 4$ Amps



$V_T = ?$
$R_T = 8\Omega$
$I_T =$ 3 amps

$V = IR$
$= (8)(3)$
$= 24$ Volts



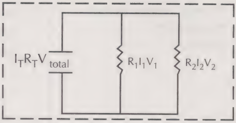
$V_T = 20$
$R_T = ?$
$I_T =$ 2 amps

$R = \frac{V}{I} = \frac{20}{2} = 10\Omega$

- **THINGS TO REMEMBER FOR PARALLEL CIRCUITS**

1. Voltage remains constant.

   $V_{total} = V_1 = V_2 \ldots$



2. Total Current = Sum of individual currents in system.

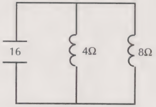
   $I_{total} = I_1 + I_2 + \ldots$

3. The lower the impedance, the greater the current demand from the system.

4. Of two resistive loads, the load with the lower impedance will receive more current (#3) and therefore more power (Watts). Power is a function of resistance and current.

---

1. Find current of each load. 4Ω (a) and 8Ω (b)
2. Find total current demand in system.
3. Find power for each load. 4Ω (a) and 8Ω (b)
4. Find total power in system.



1. $V = IR \Rightarrow I = \dfrac{V}{R}$

   Load a $I = \dfrac{16}{4} = 4$ amps   Load b $I = \dfrac{16}{8} = 2$ amps

2. $I_{total} = Ia + Ib = 4 + 2 = 6$ amps

3. Power $= \dfrac{V^2}{R}$ or I^2R

   Load a $\dfrac{16^2}{4} = \dfrac{256}{4} = 64$ Watts or $4^2 \bullet 4 = 16 \bullet 4 = 64$ Watts

   Load b $\dfrac{16^2}{8} = \dfrac{256}{8} = 32$ Watts or $2^2 \bullet 8 = 4 \bullet 8 = 32$ Watts

4. Power$_{total}$ = Pa + Pb… = 64 + 32 = 96 Watts

   or

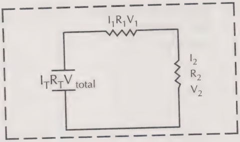
   $\text{Power}_{total} = \dfrac{V^2_{total}}{R_{total}}$ $R_{total} = \dfrac{1}{\frac{1}{4} + \frac{1}{8}} = \dfrac{8}{3}$

   $\dfrac{16^2}{\frac{8}{3}} = \dfrac{3 \bullet 16^2}{8} = \dfrac{3 \bullet 256}{8} = 3 \bullet 32 = 96$ Watts

- **THINGS TO REMEMBER FOR SERIES CIRCUITS**

1. Total Voltage = Sum of individual voltages in system.

   $V_{total} = V_1 + V_2 + V_3 \ldots$



2. Current remains constant.
   $I_{total} = I_1 = I_2 = I_3 \ \ldots$

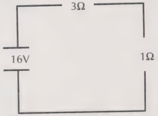
3. Resistance total = sum of individual resistors
   $R_{total} = R_1 + R_2 + R_3 \ldots$

4. Power in a series circuit. Since current remains constant...
   $P = I^2 R$

5. Of two resistive loads, the higher impedance load will get more power.

---

1. Find total current.
2. Find voltage at each load. 3Ω (a) and 1Ω (b)
3. Find power for each load.
4. Find total power in system.



1. $V_t = 16$ $\quad I_t = \dfrac{16}{4} = 4$ Amps
   $R_t = 4\Omega$

2. Load (a) $V = IR = 4 \bullet 3 = 12V$

   Load (b) $V = 4 \bullet 1 = 4V$

3. Load (a) $P = I^2R; P = 4^2 \bullet 3 = 48W$

   Load (b) $P = 4^2 \bullet 1 = 16W$

4. Total Power
   $P_t = Pa + Pb \Rightarrow 16 + 48 = 64W$
   or
   $P_t = \dfrac{V^2}{R} \Rightarrow \dfrac{16^2}{Ra + Rb} = \dfrac{16^2}{3 + 1} =$
   $\dfrac{16^2}{4} = 64W$

FORMULAS
FORMULAS
FORMULAS

• OHM'S LAW

For Parallel Circuits

$$R_t = \frac{1}{\frac{1}{R_1} + \frac{1}{R_2} + \frac{1}{R_3} \dots}$$

$$R_t = \frac{R_1 \times R_2}{R_1 + R_2} \quad \text{(for 2 resistors)}$$

For Series Circuits

$$R_t = R_1 + R_2 + R_3 \dots$$

• VOLTAGE LAW

$V = IR$ (Voltage = Current x Resistance)

$$V = IR \qquad \frac{V}{R} = I \qquad \frac{V}{I} = R$$

• POWER LAW

$P = VI$ (Power = Voltage x Current)

$$P = VI \qquad P = \frac{V^2}{R} \qquad P = I^2R$$

• REACTANCE LAW

For inductors

$$X_L = 2\pi fL$$

For capacitors

$$X_C = \frac{1}{2\pi fC}$$

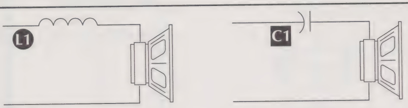
where: L is the inductance in Henries
C is the capacitance in farads
f is the frequency at the testing point

• dB LAW

$$\Delta dB = 20 \log \left(\frac{V_2}{V_1}\right) \qquad \Delta dB = 10 \log \left(\frac{P_2}{P_1}\right)$$

$$\Delta dB = 10 \log \left(\frac{V_2^2}{V_1^2}\right) \qquad \Delta dB = 20 \log \left(\frac{R_1}{R_1 + R_2}\right) \text{ (for series circuit)}$$

6dB/Octave High and Low-Pass Filters



6dB/Octave Low-Pass

6dB/Octave High-Pass

Formulas

$L = \frac{R}{2\pi f}$

$C = \frac{1}{2\pi f R}$

L = inductor in Henries
C = capacitor in farads
f = frequency
R = resistance of speaker

Table of Component Values

Frequency Hertz	Speaker Impedance					
	2 Ohms		4 Ohms		8 Ohms	
	L	C	L	C	L	C
80 100 130	4.1mH 3.1mH 2.4mH	1000µF 800µF 600µF	8.2mH 6.2mH 4.7mH	500µF 400µF 300µF	16mH 12mH 10mH	250µF 200µF 150µF
200 260 400	1.6mH 1.2mH .8mH	400µF 300µF 200µF	3.3mH 2.4mH 1.6mH	200µF 150µF 100µF	6.8mH 4.7mH 3.3mH	100µF 75µF 50µF
600 800 1000	.5mH .41mH .31mH	136µF 100µF 78µF	1.0mH .82mH .62mH	68µF 50µF 39µF	2.0mH 1.6mH 1.2mH	33µF 26µF 20µF
1200 1800 4000	.25mH .16mH .08mH	66µF 44µF 20µF	.51mH .33mH .16mH	33µF 22µF 10µF	1.0mH .68mH .33mH	16µF 10µF 5µF
6000 9000 12000	51mH 34mH 25mH	14µF 9.5µF 6.6µF	.10mH 68µH 51µH	6.8µF 4.7µF 3.3µF	.20mH .15mH 100µH	3.3µF 2.2µF 1.6µF

Note: These values are approximate for pure resistive loads only. Speakers vary in impedance with frequency and type, and will not match these values exactly.

12dB/Octave High and Low-Pass Filters

12dB/Octave Low-Pass

12dB/Octave High-Pass

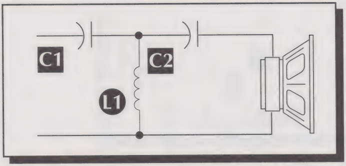
BUTTERWORTH ALIGNMENT Q = .707

Table of Component Values

Frequency Hertz	Speaker Impedance					
	2 Ohms		4 Ohms		8 Ohms	
	L	C	L	C	L	C
80 100 130	5.6mH 4.5mH 3.5mH	700µF 500µF 470µF	11mH 9.1mH 6.8mH	330µF 270µF 200µF	22mH 18mH 15mH	180µF 150µF 100µF
200 260 400	2.3mH 1.7mH 1.1mH	330µF 220µF 140µF	4.7mH 3.6mH 2.2mH	150µF 100µF 68µF	9.1mH 6.8mH 4.7mH	75µF 50µF 33µF
600 800 1000	.75mH .56mH .45mH	100µF 68µF 55µF	1.5mH 1.0mH .91mH	47µF 33µF 27µF	3.0mH 2.0mH 1.8mH	26µF 15µF 13µF
1200 1800 4000	.38mH .25mH .11mH	47µF 33µF 14µF	.75mH .50mH .22mH	22µF 15µF 6.8µF	1.5mH 1.0mH .47mH	11µF 6.8µF 3.3µF
6000 9000 12000	75mH 50mH 38mH	10µF 6µF 4.7µF	.15mH .10mH 75mH	4.7µF 3.3µF 2.2µF	.33mH .20mH .15mH	2.2µF 1.5µF 1.0µF

Note: *These values are approximate for pure resistive loads only. Speakers vary in impedance with frequency and type, and will not match these values exactly.*

18dB/Octave High and Low-Pass Filters



BUTTERWORTH ALIGNMENT Q = .707

Table of Component Values

Frequency Hertz	Speaker Impedance					
	4 Ohms			8 Ohms		
	C1	L	C2	C1	L	C2
80	330µF	6.0mH	1000µF	160µF	12mH	500µF
100	270µF	4.7mH	800µF	150µF	10mH	400µF
130	200µF	3.9mH	600µF	100µF	7.5mH	300µF
200	130µF	2.4mH	400µF	68µF	5.4mH	200µF
260	100µF	1.8mH	300µF	50µF	3.3mH	150µF
400	68µF	1.2mH	200µF	33µF	2.4mH	100µF
600	47µF	.80mH	130µF	91µF	1.6mH	68µF
800	33µF	.60mH	100µF	16µF	1.2mH	50µF
1000	27µF	.47mH	75µF	13µF	.90mH	39µF
1200	22µF	.39mH	68µF	11µF	.80mH	33µF
1800	15µF	.27mH	47µF	7.5µF	.50mH	22µF
2000	13µF	.24mH	40µF	6.8µF	.47mH	20mF
3000	8.8µF	.16mH	27µF	4.7µF	.33mH	14µF
4000	6.8µF	.12mH	20µF	3.3µF	.24mH	10µF
6000	4.7µF	82mH	13µF	2.2µF	.27mH	6.8µF
8000	3.3µF	60mH	20µF	1.5µF	.12mH	5.0µF
1000	2.7µF	47mH	8.2µF	1.3µF	.10mH	3.9µF
12000	2.2µF	39mH	6.8µF	1.1µF	82mH	3.3µF

Note: *These values are approximate for pure resistive loads only. Speakers vary in impedance with frequency and type, and will not match these values exactly.*

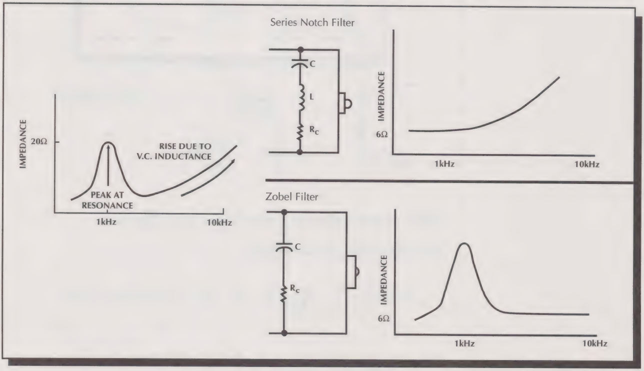
Impedance Compensation Networks

To maximize the operation of passive crossover networks and the performance of a particular driver in a multi-amplifier configuration, impedance compensating networks are very useful.

Passive crossover design is based on a static (non-changing) resistance/ impedance. Unfortunately, speakers present a dynamic load (impedance) to a crossover. Taking measurements with your IM-1 will help uncover possible impedance anomalies. With an accurate impedance plot, you can construct an impedance compensating network that is designed to minimize impedance variations at, or near the crossover frequency that will ensure that the crossover will behave as it was intended.

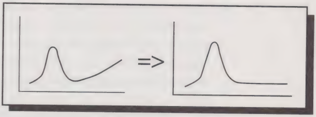
In an ***active*** system, the dynamic impedance of a driver is also very important. Many manufacturers compensate for driver anomalies with impedance compensating networks. (Some manufacturers recommend that you only use their passive crossover with their speakers.) While, at Rockford, we do not put such limitations on our speaker systems; many drivers can benefit (sound even better) from impedance networks. Remember, amplifiers produce different amounts of power (watts) depending on the load. Minimizing impedance fluctuations through the use of impedance networks can bring your sound system to the next level of sonic performance. However, the only way to construct these networks is to measure the impedance response with a device like the IM-1.

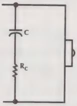
Below are some examples of impedance compensating networks.



• Zobel Network

Object: To flatten the impedance response rise caused by voice coil inductive interactions (reaction). This improves LP crossover response and reduces harshness in tweeters while providing a stable impedance platform for passive attenuation circuits (L-Pad).





$$C = \frac{L_e}{R_C^{\,2}}$$ where L_e = Driver's voice coil inductance in Henrys (not milliHenrys)

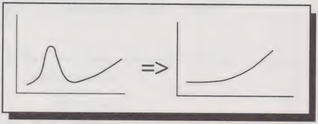
$R_C = 1.25 \times R_E$
R_E = DC resistance of speaker

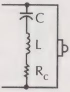
Note: *This set of values for R & C are approximate. Adjust values to achieve flat impedance response.*

Note: *1 milliHenry or 1mH = 1 x 10^{-3}H*

• Series Notch Filters

Object: To dampen and eliminate the effect that driver resonances can cause on crossover response.





$$C = \frac{0.1592}{R_e Q_{es} f_s} \ ; \qquad L = \frac{0.1592\,(Q_{es}\,R_e)}{f_s}$$

$$R_C = \left[R_e + \frac{(Q_{es}\,R_e)}{Q_{ms}} \right]$$

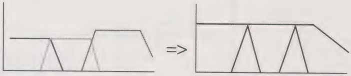
Note: *Driver parameters available from speaker factbook.*

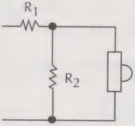
If T/S parameters are unavailable:

$$C = \frac{0.03003}{f_s} \ ; \quad L = \frac{0.02252}{f_s^{2} c} \ ; R_C = R_e \text{ (DC resistance of speaker)}$$

Note: *Measure impedance and adjust R_c in 0.5Ω steps to achieve desired impedance response.*

L-PAD ATTENUATION CIRCUIT





$$R_2 = \frac{10^{(\Delta/20)} \times Z}{1 - 10^{(\Delta/20)}}$$

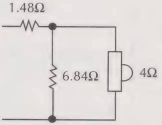
$$R_1 = Z - \left(\frac{1}{R_2} + \frac{1}{Z}\right)^{-1}$$

Where: Z = total driver impedance
A = amount of desired attenuation level in –dB
(i.e. –3dB; A = –3)

Example:
$Z = 4\Omega$; $A = -4$ (–4dB)

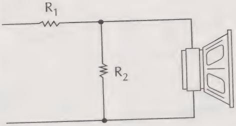
$$R_2 = \frac{10^{(-4/20)} \times 4}{1 - 10^{(-4/20)}} = \frac{10^{(-1/5)} \times 4}{1 - 10^{(-1/5)}} = 6.84\Omega$$

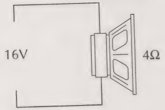
$$R_1 = 4 - \left(\frac{1}{6.84} + \frac{1}{4}\right)^{-1} = 1.48\Omega$$



L-Pad Example

W = 64 Watts



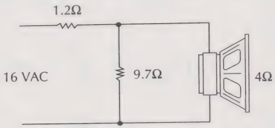


Goal: Reduce acoustical output by 3dB (32 Watts power applied).

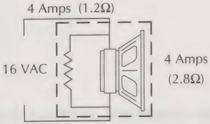
$$R_2 = \frac{10^{(A/20)} \bullet Z}{1 - 10^{(A/20)}} \qquad R_1 = Z - \left(\frac{1}{R2} + \frac{1}{Z}\right)^{-1}$$

$$R_2 = \frac{10^{(-3/20)} \bullet 4}{1 - 10^{(-3/20)}} = 9.696\Omega \approx 9.7\Omega$$

$$R_1 = 4 - \left(\frac{1}{9.696} + \frac{1}{4}\right)^{-1} = 1.168\Omega \approx 1.2\Omega$$

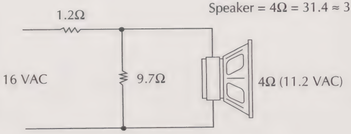


Total Ω = 4 Ohms
Total W = 64 Watts

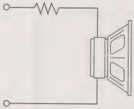


$I_t = I_1 = I_2 = 4$ Amps

R_1 = 1.2Ω, 4 Amps, 19.2 Watts, 4.8 Volts
R_2 = 9.7Ω, 12.9 Watts
Speaker = 4Ω = 31.4 ≈ 32 Watts



• **Series Resistor Attenuation Circuits**



$$Rc = \frac{Re\,(1 - 10^{\frac{A}{20}})}{10^{\frac{A}{20}}}$$

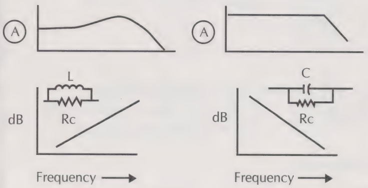
where: R_c = Attenuation resistor
R_e = DC resistance of voice coil
A = dB attenuation i.e., –3dB A = –3

to find dB of attenuation…

$$A\,(dB) = 20 \log \left(\frac{Re}{Re + Rc}\right)$$

• **Frequency Response Shaping Network**

Contour Networks



Example 1: Response is rising with increasing frequency

$$L = \frac{0.15916}{f} \qquad R_C = \frac{Re\,(1-10^{\frac{A}{20}})}{10^{\frac{A}{20}}}$$

$$z = \frac{R_C X_L}{(R_C^2 + X_L^2)^{1/2}}$$

L = inductance in Henries
f = starting point of network interactance
z = total circuit impedance
Re = resistance of voice coil
A = amplitiude of signal
–3dB = –3 (for A)

$X_L = 2\pi fL$ formula for reactance of an inductor
f = frequency of maximum desired attenuation
X = component reactance at frequency of maximum attenuation

$$A_t = 20 \log \frac{Rd \,\&\, Z}{Rd}$$

Rd = total driver impedance

Example 2: Response is rising with decreasing frequency

$$c = \frac{0.15916}{f}$$

$$z = \frac{R_C X_C}{(R_C^2 + X_C^2)^{1/2}}$$

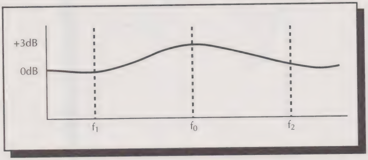
c = capacitance in farads
f = starting point of network interactance
z = total circuit impedance

$X_C = \dfrac{1}{2\pi fc}$ formula for reactance of an inductor
f = frequency of maximum desired attenutation
X = component reactance at frequency of maximum attenuation

$$A_t = 20 \log \frac{Rd \,\&\, Z}{Rd}$$

Rd = total driver impedance

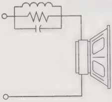
Design Concept: Network designed to compensate for frequency peaks caused by driver response and vehicle acoustics.



Step 1: Find f_0

Step 2: Find f_1 & f_2 (–3dB point referenced to f_0)

Step 3: Design Circuit



$$C = \frac{0.03003}{f_0} \quad \text{in farads}$$

$$L = \frac{0.02252}{f_0^2 C}$$

$$B = \text{–3dB Bandwidth} = (f_2 - f_1)$$

$$R = \frac{1}{6.2832\ CB}$$

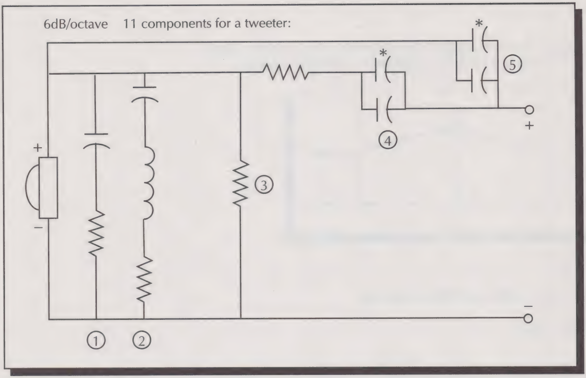
Example: f_0 = 1000Hz; f_2 = 2000Hz; f_1 = 500Hz

$$C = \frac{0.03003}{f_0} = 30.03\mu f$$

$$L = \frac{0.02252}{(1000)^2\,(3.003 \times 10^{-5})} = 0.75\text{mH}$$

$$B = 2000 - 500 = 1500$$

$$R = \frac{1}{6.2832\,(3.003 \times 10^{-5})\,(1500)} = 3.5\Omega$$



Types of passive filters used:

1. **Zobel Network**
   Compensates for impedance rise due to inductance of voice coil.

2. **Series Notch Filter**
   Compensates for impedance rise due to resonance of driver.

3. **L-Pad Network**
   Compensates for sensitivity differences between tweeter & midrange.

4. **Crossover**
   High-pass crossover passes high frequencies while blocking low frequency information.

5. **Frequency Shaping Contour Network**
   Network compensates for increasing frequency response roll-off due to off-axis frequency response of the driver.

* Bypass capacitor combines with other capacitor to provide greater performance of a capacitor at a much reduced cost.

CHAPTER 8

ACTIVE COMPONENT SYSTEMS

- XCARD ORIENTATION
- XCARD CONSTRUCTION
- USING XCARD RESISTOR CHARTS
- DETERMINING CROSSOVER FREQUENCY OF PRE-FABRICATED XCARDS
- BUTTERWORTH ALIGNMENTS
- HIGH-PASS FILTERS FOR LOW FREQUENCY SYSTEMS
- BUTTERWORTH ALIGNMENT SUB-SONIC FILTERS
- HIGH "Q" HIGH-PASS FILTERS
- DESIGNING HIGH "Q" FILTERS
- BUILDING HIGH "Q" FILTERS
- MAPPING TRANSFER FUNCTION OF XCARDS

Active Component Systems

XCard Orientation

The XCard™ can be set for high-pass, low-pass, or full range operation. Prefabricated XCards are available from Rockford Fosgate in the most common frequencies. Each XCard (has two faces: one face operates **Full Range**, the other has arrows to indicate the edge for selecting **HP** (high-pass) or **LP** (low-pass) operation. Orient the card with the desired operating edge, indicated by the arrow, toward the socket terminals (the metal fingers in the crossover slot) inside the amplifier or crossover. Firmly, but carefully, plug the XCard into to socket. *(Figure 8.1)*

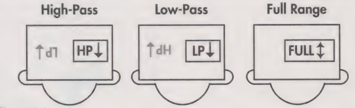


Figure 8.1

XCard Construction

One of the many benefits of the XCard is the ability to construct custom cards at any frequency or alignment. This is accomplished by changing the resistor values on the card itself. To change and place the resistors on the Cards refer to the *Placing Resistor* section in Step 3.

• Placing Resistors

There are two ways to begin this task. One using an XM00 XCard and the other is to modify an existing XCard. Either procedure is easily accomplished using the following steps.

Tools Needed

Soldering Iron
Solder Wick or Solder Sucker
Solder
Wire Cutters
Volt/Ohm Meter (Optional)

Step 1

Remove existing resistors. Using a soldering iron and solder sucker or wick, remove the resistor being careful not to apply heat to the capacitors. Heat applied to the capacitors will damage them. This step is not necessary with the XM00 XCard.

Step 2

Determine and verify the resistor values needed. Refer to the following sections to determine the proper resistor values needed. In addition, it is always beneficial to measure the resistors with a Volt/Ohm meter to ensure crossover frequency accuracy.

STEP 3 **Place resistors on the XCard.** Place the resistors (R1 and R2) in the second, fourth, sixth and eighth positions on the XCard side that has the "R" in the upper left portion of the card. Refer to *Figure 8.2.*

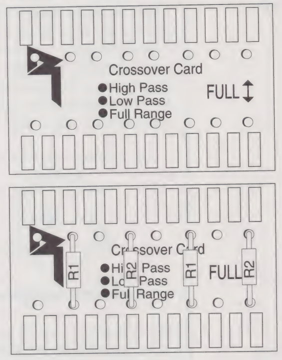


Figure 8.2

STEP 4 **Solder the resistor.** Solder the resistors to the XCard being careful not to apply too much heat either to the circuit board or the capacitors as this will damage them. For best results use a solder iron with a 600° to 800°F fine tip.

STEP 5 **Insert the XCard into the amplifier or crossover.** Insert the XCard into the amplifier or crossover as described in the XCard Orientation section of this chapter. ***Warning! Turn the system off when changing the XCard in your system.***

HELPFUL HINT Remember to practice good solder techniques when removing and adding resistors. Clean any extra solder with a solder wick to prevent the XCard from shorting when placed in the crossover slot. **Failure to do this can cause speaker and system damage.**

Using XCard Resistor Charts

Rockford Fosgate supplies a quick reference charts for resistor values for both Butterworth Alignments and Bessel Alignments. To use the XCard Resistor Charts, follow the steps below.

Note: The XCard Resistor Charts are located in the Appendix of this manual.

Step 1

Determine the capacitor value of the XCard. The XCard is manufactured with both 0.022µF capacitors and 0.047µF capacitors. This value will determine the following steps needed.

Step 2

For different value capacitors use the following instructions below.

For 0.022µF capacitors, simply use the chart.

For 0.047µF Capacitors, take the resistor values and multiply the amount by 0.47 to convert for use for the 0.047µF capacitor XCard.

Step 3

Place Resistors on the XCard. When building alignments other than Butterworth ("Q" other than 0.707), the XCard used can only be a dedicated high-pass or low-pass crossover card. The resistor values needed for the other alignments are not compatible in both configurations. Therefore, it is important to place the resistor values R1 and R2 in the correct location. Refer to the *Placing Resistors* section in this chapter.

Step 4

Insert the XCard into the amplifier or crossover. Insert the XCard into the amplifier or crossover as described in the *XCard Orientation* section of this chapter. ***Warning! Turn the system off when changing the XCard in your system.***

Determining Crossover Frequency of Pre-fabricated XCards

Determining the crossover frequency value of the XCards can be accomplished by using a variation of the formulas used to calculate resistor values for Butterworth Alignments. On pre-fabricated XCards, the values for Resistor 1 and Resistor 2 are the same. Due to this feature, if all the resistors on the XCard are the same value, the XCard is a Butterworth alignment. Listed below are the steps to determine the crossover frequency of pre-fabricated XCards.

Step 1

Determine the capacitor value used on the XCard. The XCard is manufactured using either 0.022μF or 0.047μF capacitors. To determine their value either read the numbers listed on the capacitor (223 for 0.022μF or 473 or 0.047μF) or measure the value of the capacitor with either a capacitance meter or a multimeter that has the ability to measure capacitance. (Fluke 12)

Step 2

Use the following formulas:

For 0.022μF Capacitors

Crossover Frequency, $f_o = \frac{7234}{R}$ $R = R_1$ and R_2 kΩ

For 0.047μF Capacitors

Crossover Frequency, $f_o = \frac{3386}{R}$ $R = R_1$ and R_2 kΩ

Where:
- f_o = Crossover frequency of the XCard
- C = Capacitor value used on the XCard
- R = The resistance of R_1 and R_2 (R_1 and R_2 must be the same value)
- R_1 = The resistor in the 1st and 3rd resistor position on the XCard
- R_2 = The resistor on the 2nd and 4th resistor position on the XCard

Example:

Given information:
C = 0.047μF
R_1 = 1kΩ
R_2 = 1kΩ
R = 1kΩ

Calculations:

Crossover Frequency, $f_o = \frac{3386}{R} \Rightarrow \frac{3386}{R} \Rightarrow$ 3386Hz

BUTTERWORTH ALIGNMENTS

Butterworth Alignments differ from other alignments possible with the XCard. The values for both Resistor 1 and Resistor 2 are the same for both high-pass and low-pass alignments. Due to this feature, one constructed Butterworth alignment XCard can be used either as a high-pass or low-pass crossover card at that particular crossover frequency. Listed below are the steps to determine the proper resistor values for Butterworth Alignments.

STEP 1 **Determine the capacitor value of the XCard.** The XCard is manufactured with both 0.022µF capacitors and 0.047µF capacitors. This value will determine which of the following formulas to calculate the resistor values.

STEP 2 **Use the following formulas:**

For 0.022µF Capacitors

$$R_1 = R_2 = \frac{7234}{f_0} \qquad R_1 \text{ and } R_2 \text{ are in k}\Omega$$

For 0.047µF Capacitors

$$R_1 = R_2 = \frac{3386}{f_0} \qquad R_1 \text{ and } R_2 \text{ are in k}\Omega$$

STEP 3 **Place Resistors on the XCard.** For detailed instructions on placing the resistors on the XCard refer to *Placing Resistors* in this section.

STEP 4 **Insert the XCard into the amplifier or crossover.** Insert the XCard into the amplifier or crossover as described in the *XCard Orientation* section of this chapter. ***Warning! Turn the system off when changing the XCard in your system.***

Example:
Given information
Desired Crossover Frequency: 40Hz
Capacitor Value: 0.047µF

Calculations

$$R_1 = R_2 = \frac{3386}{f_0} \Rightarrow \frac{3386}{f_0} \Rightarrow 84.65\text{k}\Omega \approx 84.7\text{k}\Omega$$

Note: When exact values are not available, two resistors in series can be used. The above example can be accomplished by using a 82,000Ω resistor in series with a 2,700Ω resistor for a total of 84,700Ω.

High-Pass Filters for Low Frequency Systems

Many understand the need for crossovers. In an ideal system only one speaker would be needed to reproduce the entire audio bandwidth from 20Hz to 20kHz. With today's technology, this is impossible. All conventional speakers have operational bandwidths. We group a speaker into a family (Woofer, Mid-bass, Midrange, Tweeter) depending of the frequency range that particular speaker can play effectively. It is possible that a speaker could cover a large enough bandwidth and could be placed in more than one speaker family designation.

• High-Pass for a Woofer?

High-pass crossovers are often overlooked for low frequency drivers. The need for high-pass crossovers on a tweeter in obvious. Below a certain frequency, a tweeter does not have very much output. If the tweeter is driven at these lower frequencies to try to reproduce these lower frequencies. Poof! No more tweeter.

What is the difference between a tweeter and a woofer? The answer is only the frequency range it was designed to optimize. Below a certain frequency, a woofer does not have very much output. If the woofer is driven at these lower frequencies – Poof! – No more woofer.

To prevent damage to the tweeter, we use a high-pass filter. But what about the woofer? Using a high-pass crossover on a woofer will improve power handling, increase reliability, and most of all, improve sonic performance. The high-pass filter for a woofer is also called a Sub-Sonic Filter. In this section are applications of high-pass crossovers for low frequency speakers.

Butterworth Alignment Sub-Sonic Filters

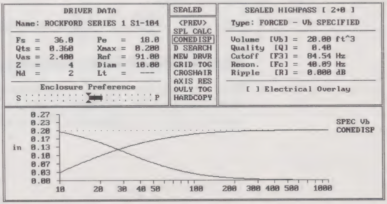
One example we will use is the installation of a woofer in an infinite baffle (very large) enclosure. The woofer is a Series 1 10" speaker. The infinite baffle will consist of a baffle board mounted to the rear deck of a vehicle with a trunk. The trunk of the vehicle is average size. The design goal is to maximize the amplifier power the speakers can handle without exceeding the X_{max} or the maximum excursion of the speakers.

Example

Woofers :	Series 1 S1-104
Enclosure Type :	Infinite Baffle
Volume of enclosure:	20 ft^3 or 566 liters

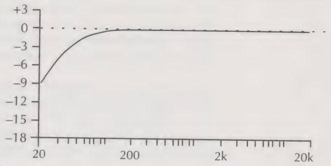
- 75 Watts of Power Handling at any frequency
- No perceptible change in frequency response

• Frequency and Excursion Plots



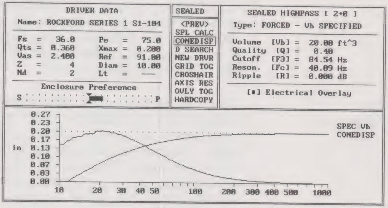
- Only 18 Watts of Power Handling at 10Hz

- **XCard Sub-Sonic Filter**

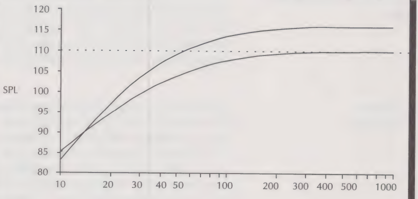


• High-Pass Crossover applied at 40Hz, the F_C of the enclosure

- **Frequency and Excursion Plots with Sub-Sonic Filter Applied**



- No perceptible change in frequency response
- Large gain in perceived sound quality



• Difference in SPL with and without High-Pass Crossover.

The result is a woofer with excellent power handling and improved reliability. The amplifier of choice would either be a Series 1 2600x or a Punch 60ix (because of their XCard capability).

High "Q" High-Pass Filters

In the past, RTTI most often recommended tuning a bass reflex enclosure between 40-45Hz. This enclosure design offered excellent performance for the time it was designed. During that era, most music software available did not have much sonic information below 45Hz. In fact, the bass portion of the music mostly centered around 60Hz.

Because the tuned frequency of the enclosure centered between 40-45Hz, boost from the Punch EQ could be applied, increasing the output of that enclosure. The bass reflex enclosure tuned to 45Hz and the patented Punch EQ boost capability centered at that same frequency combined to make a winning combination.

• Enclosure Design

A perfect enclosure design does not exist. All enclosures have benefits and disadvantages. Bass reflex designs, also referred to as vented or ported enclosures, are no different in this respect. At and above the tuned frequency, bass reflex enclosures benefit from an increase in efficiency when compared to acoustic suspension (sealed) enclosures of similar size. The drawback of bass reflex enclosures is reduced mechanical power handling for the woofer and steeper frequency response roll-off below the tuned frequency.

• Old Designs, New Challenges

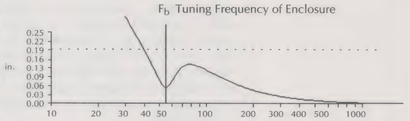
That was great for the past, but what about today? Today we have music with information well below 45Hz. Enclosure designs, considered awesome in the past, are not always capable for the musical demand of today. Mechanical power handling of the woofer greatly diminishes below the tuned frequency of a bass reflex enclosure. Bass reflex enclosures do not offer any assistance to the woofer's mechanical power handling below the tuned frequency. In effect, the woofer would behave as if it is mounted in air, hanging only by a string. If the customer were to attempt to play that speaker at lower frequencies, at high SPLs, the result would be another trip to the Rockford Dealer to replace a blown woofer.

• Defining Performance Goals

What we need is a solution for today. To accomplish this, we must understand all the important factors that define the performance goals we want to achieve. We need to look at enclosure, software material available, and the audio system including the equalizer, crossover and amplifier we want to use to drive the woofer in the enclosure.

• Excursion Excursion

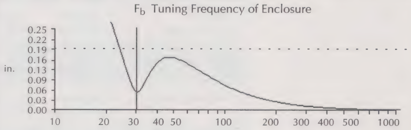
Woofers in a bass reflex enclosure reach a minimum cone excursion at the tuned frequency of the enclosure. This is caused by the interaction of the speaker, port and the enclosure. At the tuned frequency of the enclosure, the majority of the output of the enclosure is produced by the port. Mechanical power handling of a woofer is very high due to its minimum cone movement at that tuned frequency. *Figure 8.3* displays the cone displacement plot to Term-Pro. The graph represents cone displacement as a function of frequency and power applied. The vertical axis represents the cone movement versus input power and the horizontal axis displays frequency.



- F_b or Tuned Frequency of the enclosure is 45Hz
- Above the F_b, cone excursion is well controlled
- Below the F_b, cone excursion exceed the woofer 's capability at 35Hz
- Peak excursion above the F_b is 62Hz
- Mechanical power handling extends to 35Hz
- The dotted line represents the X_{max} capability of the speaker

Figure 8.3

The dotted line represents the maximum excursion, X_{max} capability of the speaker. Exceeding this line represents exceeding the mechanical limitations (power handling) of the speaker. Notice at the tuned frequency of the enclosure the power handling capabilities of the speaker are very good. In this application, the Punch EQ could be used to increase the 45Hz range of bass output. Unfortunately, being concerned with the power handling of the speaker around 30Hz, this enclosure probably would not meet the needs of our performance goals. The solution would appear to simply lower the tuned frequency. But would the Punch EQ still be effective in this application? *Figure 8.4* is the same woofer with the same amount of power applied. The only difference is that the tuned frequency is lowered to 30Hz. Several things happen due to this circumstance.



- F_b or Tuned Frequency of the enclosure is 30Hz
- Above the F_b, cone excursion is increased over the 45Hz tuning
- Below the F_b, cone excursion increases at a faster rate when compared to the 45Hz tuning
- Peak excursion above the F_b is 45Hz
- Mechanical power handling extends to 23Hz
- The dotted line represents the X_{max} capability of the speaker

Figure 8.4

Due to the excursion peak at 45Hz, the Punch EQ would not be recommended. Although the enclosure has greater peak excursion above the tuned frequency, the enclosure exhibits good mechanical power handling in the bandwidth the speaker would be driven. Using the Punch EQ with this enclosure would easily exceed the X_{max} of the speaker. Even though the Punch EQ is no longer effective, the concept of boosting at the tuned frequency of the enclosure is a useful way of increasing low frequency extension. How can we incorporate this with the Punch EQ?

- **High "Q" Solution**

One of the most unique benefits of the XCard is the ability to use any alignment or "Q". A low "Q" (below 0.707) has a dip in the response before the roll-off. A high "Q" alignment (above 0.707) has boost in the response before the roll-off. If the crossover is a high "Q" high-pass alignment, the peak frequency of the crossover can be centered at the tuned frequency of the low frequency woofer system.

- **Maximum Boost Available**

The maximum boost the XCard is capable of is greater than 17dB at frequencies lower than 20Hz. This capability is obviously greater than the capability of any low frequency enclosure design and any amplifier's dynamic ability. The practical boost limit is about 10dB. This is equal to ten times the output required from the amplifier. When is this boost usable? Different styles of enclosure and woofer combinations can accept different boost levels. For the maximum usable boost for any particular enclosure refer to *Table 8.1.*

Boost Levels	**0dB Q = 0.707**	**+3dB Q = 1.4**	**+6dB Q 2.0**	**+10dB Q = 3.5**
Enclosure Type Infinite Baffle	Acceptable No Problems	Use Term-Pro	Not Acceptable	Not Acceptable
Sealed	Acceptable No Problems	Acceptable No problems "Q" =1.1	Use Term-Pro	Not Acceptable
Vented No Problems	Acceptable No Problems	Acceptable No Problems	Acceptable Term-Pro	Use
Sealed Bandpass	Acceptable No Problems	Acceptable No Problems	Use Term-Pro	Not Acceptable
Vented Bandpass	Acceptable No Problems	Acceptable No Problems	Acceptable No Problem	Use Term-Pro
Aperiodic	Set crossover frequency to f_s of speaker	Set crossover frequency to f_s of speaker	Set crossover frequency to f_s of speaker	Not Acceptable

Table 8.1

In some applications, the performance needs to be verified by Term-Pro. These boost levels are near the peak performance capabilities of the combination of the enclosure and the speaker. If Term-Pro or another capable software program cannot be used, do not attempt filters at these boost levels.

DESIGNING HIGH "Q" FILTERS

- **INFINITE BAFFLE FILTERS**

When designing a filter for an infinite baffle enclosure, mechanical power handling and low frequency performance are the two important criteria to consider. For 0dB alignments, selecting the high-pass crossover frequency at the F_c of the enclosure will typically operate the best. If any boost is desired, use Term-Pro to determine the mechanical power handling of the speaker.

- **SEALED ENCLOSURE FILTERS**

Sealed enclosures operate very similar to infinite baffle enclosures. In fact, infinite baffle enclosures are just very large sealed enclosures. When designing 0dB alignments, select a high-pass filter that corresponds with the resonant frequency (F_C) of the enclosure or the point where the output of the enclosure is reduced by 3dB or the F_3 of the enclosure, which ever is lower. When using boost in the alignments, woofers with high excursion capabilities (X_{max} equal to or greater than 0.30 inches) can usually accept boost applied in the lower frequency range. Typically sealed enclosures designed with a total "Q" around 1.1 can have boost applied into the lower frequency range without much mechanical problems. This, of course, does not always mean that the woofer can have the mechanical power handling (boost applied) equal to the thermal power handling (heat capacity of the voice coil, voice coil former and the associated glues and connections) stated in the product section of this manual. When using 6dB or more of boost to a sealed woofer, refer to Term-Pro to determine the mechanical power handling of the speaker.

- **BASS REFLEX ENCLOSURE FILTERS**

As in the examples, bass reflex enclosure mate very well to high "Q" high-pass filters. When designing these filters, it is very important to situate the peak boost of the filter at the tuned frequency of the enclosure. Obviously, at 0dB of boost selecting the tuned frequency for the crossover frequency poses no problems. At higher levels of boost, enclosure volume will affect the mechanical power handling of the speaker in the enclosure. The larger the enclosure, the lower the mechanical power handling. The smaller the enclosure volume, the greater the mechanical power handling. The inverse is true of low frequency efficiency. The larger the enclosure, the less power is required to reproduce low frequency. When using large amounts of boost, it is necessary to use a speaker with a large excursion capability and verify the excursion in the enclosure with Term-Pro.

- **SEALED AND VENTED BANDPASS ENCLOSURE FILTERS**

Bandpass enclosures operate in much the same way as their more basic counterparts. Refer to the preceding sections of this chapter and use the same design criteria. Sealed bandpass design criteria function similar to sealed enclosures. Vented bandpass enclosures function similar to vented enclosures in the low frequency range.

- **OBEY MOTHER NATURE**

Other factors determine the capabilities of the high "Q" filter. The maximum excursion of the speaker is one of the most important parameters. If all other parameters are equal, woofers with large excursion capabilities can accept more boost than woofers with less. If any of the parameters are changed, however, this statement cannot be made.

Reliability of the low frequency system depends greatly on the level setting process and the peak filter frequency relative to the enclosure. Following some basic guidelines will yield reliable performance. Disregard the laws of physics and Mother Nature will haunt you in your dreams. Some basic guidelines to follow when using high-pass high "Q" filters are on the following page.

• High "Q" Guidelines

1. If unsure about your capabilities, don't attempt these filters.
2. In all applications, 0dB of boost will never reduce the reliability of the system. It can only reduce the output capability. Correctly choosing the filter frequency will yield excellent reliability with inaudible loss in low frequency output.
3. 3dB boost applications can be used in most low frequency designs. In vented alignments, do not boost below the tuned frequency of the enclosure. In sealed alignments, do not align the filter frequency at the F_c of the enclosure.
4. 6dB boost filters for sealed alignments exceed most woofer capabilities. Remember, asking the woofer to provide 6dB more output is asking the woofer to move twice as much. There are certain woofers with very high X_{max} (like the RFA-412 and RFA-812) with the ability to accept these higher boost levels of 6dB. It is most important to use Term-Pro to verify these particular conditions for your enclosure design.
5. 6dB boost filters for vented alignments also follow the 3dB applications. But, if the woofer's X_{max} is not adequate, this alignment would not be recommended.
6. **10dB boost filters are only recommended for vented alignments.** These boost levels exceed the capability of most woofers. Woofer with good X_{max} capabilities will most often perform better at these high boost levels. It is most important to use Term-Pro to verify these particular conditions for your enclosure design.
7. When designing higher boost levels, enclosure size affects the mechanical power handling of the speaker. If a particular vented alignment design does not have the power handling required, decrease the internal volume of the enclosure. Remember, when you decrease the internal volume of the enclosure, the port length required increases dramatically.
8. In sealed enclosures, decreasing the internal volume of the enclosure has to be balanced with the low frequency output capability of the design. Often, to acquire the power handling desired, the result is an enclosure that can be outperformed with a larger enclosure powered by a smaller amplifier.

Building High "Q" Filters

The following steps will help you design the high "Q" filters. Often, the steps will ask you to refer to the *High "Q" Guidelines* in this chapter. Whenever possible, use Term-Pro to verify your calculations and results.

Step 1 — **Determine performance goals.** Before designing a filter, a plan of attack is needed to begin this journey.

Step 2 — **Determine the Crossover Frequency.** After deciding which enclosure style to be used, determine the best crossover frequency for your performance needs. Refer to the *Designing High "Q" Filters* section in the chapter for detailed information.

Step 3 — **Choose the boost the filter will use.** Refer to *Table 8.1* for detailed information on maximum boost allowable.

Step 4 — **Calculate the resistor values needed for the filter.** Refer to *Table 8.2* for the correct formula for your particular design.

Boost	0.022µF		0.047µF+10dB	
+10dB Q=3.5	$R_1 = \frac{4480}{f_o}$	$R_2 = \frac{11674}{f_o}$	$R_1 = \frac{2107}{f_o}$	$R_2 = \frac{5487}{f_o}$
+6dB Q=2.0	$R_1 = \frac{4940}{f_o}$	$R_2 = \frac{10587}{f_o}$	$R_1 = \frac{2322}{f_o}$	$R_2 = \frac{4976}{f_o}$
+3dB Q=1.4	$R_1 = \frac{5432}{f_o}$	$R_2 = \frac{9628}{f_o}$	$R_1 = \frac{2553}{f_o}$	$R_2 = \frac{4525}{f_o}$
0dB Q=.707	$R_1 = \frac{7234}{f_o}$	$R_2 = \frac{7234}{f_o}$	$R_1 = \frac{3386}{f_o}$	$R_2 = \frac{3386}{f_o}$

Note: R_1 and R_2 values are in kΩ.
f_o is the peak boost frequency of the filter.

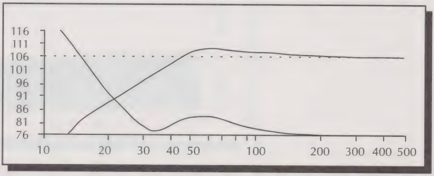
Table 8.2

STEP 5 **Place Resistors on the XCard.** For detailed instructions on placing the resistors on the XCard, refer to *Placing Resistors* in this section.

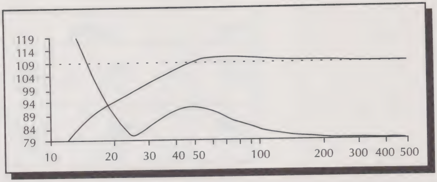
STEP 6 **Insert the XCard into the amplifier or crossover.** Insert the XCard into the amplifier or crossover as described in the *XCard Orientation* section of this chapter. ***Warning! Turn the system off when changing the XCard in your system.***

• HIGH "Q" EXAMPLE

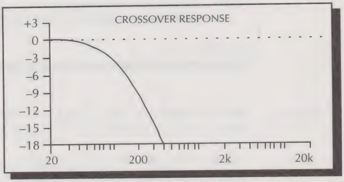
The maximum enclosure volume available is 2 cubic feet or 56.63 liters. Below are the steps used in designing the enclosure using the high-pass high "Q" filter.



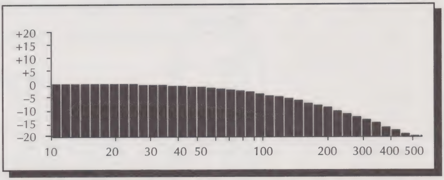
- Term-Pro Spec V_b enclosure at 2 cubic feet (56.64 liters)
- 50 Watt power handling at 15Hz
- F_3 40Hz



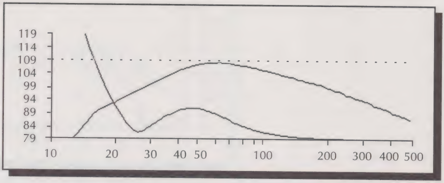
- 2 cubic foot (56.64 liters) enclosure re-tuned to 25Hz.
- 100 Watt power handling at 15Hz.
- F_3 still at 40Hz.



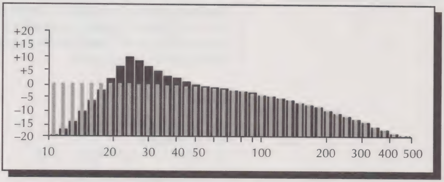
- Low-pass crossover frequency at 100Hz.



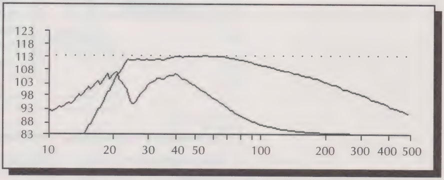
- Overlay section of Term-Pro. Low-pass crossover activated.



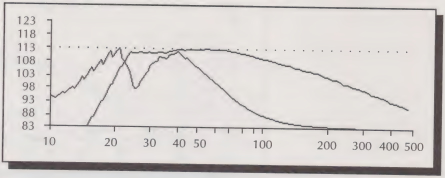
- Spec all enclosures with low-pass crossover activated.
- 125 Watt power handling at 15Hz.
- F_3 of enclosure is 40Hz.



- High-pass High "Q" filter at 25Hz with 10dB of peak boost.
- 100Hz low-pass crossover activated.



- Spec all enclosures 2 cubic feet tuned to 25Hz.
- With 250 watts, no mechanical limitations.
- F_3 of the enclosure is 22Hz.



- 400 Watts power handling at any frequency.
- F_3 of the enclosure is 22Hz.

MAPPING TRANSFER FUNCTION OF XCARD

To accurately track the operation of the XCard, the frequency response of the XCard has to be measured. Measuring the frequency response can be done with tools available around the installation bay and sales floor. Once the filter's response is plotted, the information can be imported in Term-Pro and the mechanical power handling of the speaker can be examined and optimized.

TOOLS NEEDED

OSC2
CD Source Unit
AC Voltmeter
Assorted RCA and Power Cables
Active Crossover or Amplifier
12 Volt DC Power Supply

STEP 1

Set up test bench. *Figure 8.5*

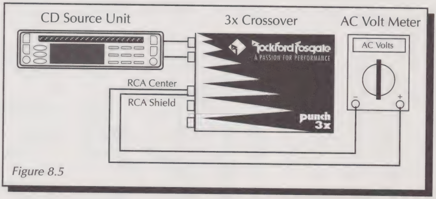


Figure 8.5

STEP 2

Insert crossover card in the amplifier or crossover in the full range position.

STEP 3

Play test signal (sine wave) from your source unit. Frequencies from 10-98Hz are available from the Autosound 2000 Test CD #101. Frequencies above 98Hz can be used but an OSC2 is needed. These frequencies on the worksheet correspond with the frequencies in the overlay section of Term-Pro.

STEP 4 **Connect an AC Volt meter to the output of the amplifier or active crossover used in the test.** With amplifiers, place the positive lead of the meter on the positive terminal of the speaker lead output. Place the negative lead of the meter on the speaker ground of the amplifier. With crossovers, place the positive lead on the center or positive RCA lead of the crossover output. Place the negative lead of the meter on the RCA shield ground of the crossover output.

Measure the Full Range Output Voltage of the amplifier or crossover. Use the meter in the AC Voltage setting. For crossovers, measure the voltage of the output in AC milliVolts. A good reference setting is around 100 mV. For amplifiers, measure the voltage in volts. A good reference voltage is 1/8 of the maximum AC Voltage output of the amplifier. (*Example:* If the amplifier's maximum AC Volts output per channel is 20 Volts, a good reference voltage would be around 1-2.5 volts) This is done to ensure enough headroom for the boost capability of the high-pass crossover. Leave the setting for the voltage at this level for the remainder of the test.

STEP 6 **While selecting the frequencies in the worksheet, record the voltage of the component in the full range mode.** For best results, start at the lowest frequency, 11Hz, and work your way through the worksheet.

STEP 7 **Insert the crossover in the high-pass position.**

STEP 8 **While selecting the frequencies in the worksheet, record the voltage of the component in the high-pass mode.**

STEP 9 **Using the formula, calculate the dB difference in output between the full range and high-pass positions.**

The result of this information can be imported into Term-Pro. This will give the ability to model the excursion of a particular enclosure, tuning frequency and woofer selection. For more detailed information regarding the Overlay section of Term-Pro refer to the *Term-Pro Operation and Application Manual.*

Advanced RTTI

High-Pass High "Q" Crossover Worksheet

Frequency: ______________________ Capacitor 1: ______________ Resistor 1: ________________

dB Boost: ______________________ Capacitor 1: ______________ Resistor 2: ________________

Formula for Δ **dB** $\Delta dB = 20\log\left(\frac{\text{High-Pass Output Voltage}}{\text{Full Range Output Voltage}}\right)$

Frequency in Hertz (Hz)	**Full Range Output Voltage** of amplifier or crossover in the full range position	**High-Pass Output Voltage** of amplifier or crossover in the high-pass position	Δ **dB** difference between the voltages with and without the crossover engaged
11 Hz			
12 Hz			
13 Hz			
14 Hz			
16 Hz			
18 Hz			
19 Hz			
22 Hz			
24 Hz			
26 Hz			
29 Hz			
32 Hz			
36 Hz			
40 Hz			
44 Hz			
49 Hz			
54 Hz			
60 Hz			
66 Hz			
74 Hz			
81 Hz			
90 Hz			

Advanced RTTI
High-Pass High "Q" Crossover Worksheet

Frequency: ______________________ Capacitor 1: ______________ Resistor 1: ________________

dB Boost: ______________________ Capacitor 1: ______________ Resistor 2: ________________

Formula for Δ **dB** $\Delta dB = 20\log\left(\frac{\text{High-Pass Output Voltage}}{\text{Full Range Output Voltage}}\right)$

Frequency in Hertz (Hz)	**Full Range Output Voltage** of amplifier or crossover in the full range position	**High-Pass Output Voltage** of amplifier or crossover in the high-pass position	Δ **dB** difference between the voltages with and without the crossover engaged
11 Hz			
12 Hz			
13 Hz			
14 Hz			
16 Hz			
18 Hz			
19 Hz			
22 Hz			
24 Hz			
26 Hz			
29 Hz			
32 Hz			
36 Hz			
40 Hz			
44 Hz			
49 Hz			
54 Hz			
60 Hz			
66 Hz			
74 Hz			
81 Hz			
90 Hz			

Chapter 9

Test Equipment

- Features & Benefits
- PD1/PD2 (Pulse Generator & Detector)
- IM-1 (Impedance Meter)
- OSC2 (Sine Wave/Pink Noise Generator)
- Volume of an Irregular Shape
- Low Frequency Test CD #101

Test Equipment

PD1 / PD2

Feature	Procedure	Benefit
Determine speaker and wire orientation.	Use the PD-1 to pulse speaker wires to locate speakers.	Safe method for locating speakers in the vehicle.
Determine OEM wiring.	Use the PD1/PD2 to find the location and polarity of a fatory stereo system.	Safe and simple method to determine speaker wiring in an OEM stereo system.
Determine polarity of speakers in a system.	Use the PD1/PD2 to determine speaker polarity.	Safe and simple method to use and will work through any acoustically transparent material such as OEM carpet.
Determine absolute polarity in a stereo system.	Using CD #101 (Track #3) measure the absolute polarity at any speaker from the source unit.	Only method to determine absolute polarity of a system.

IM-1

Feature	Procedure	Benefit
Troubleshoot system.	Measure the impedance of the speakers in the vehicle.	Ensure that system wires are not short circuited or shorted to the chassis of the vehicle. Ensure speakers have a well behaved impedance response.
Measure the impedance of a low frequency system.	Plot impedance to determine the best wiring configuration for maximizing amplifier output.	Optimizes use of power in a system.
Measure impedance of a speaker.	Plot impedance of a speaker and build compensation network to smooth impedance response.	Provides stable impedance for well behaved crossover response.

Feature	Procedure	Benefit
Find tuning frequency of a vented enclosure.	Measure actual tuning frequency of enclosure and adjust to desired tuning frequency.	Ensures maximum performance from a vented system.
Troubleshoot system.	Check and verify operation of equipment by substituting known working isolated source.	Easy method of checking operation of all components, cable and connections.
Troubleshoot system noise.	Substitute component in question with known working, noise-free, isolated source.	Easy method for verifying noise source in system.
Find resonances of vehicle.	Sweep vehicle with sine wave output and find the frequencies that cause rattles and buzzes and damp these unwanted noises.	Quick, easy method of locating and eliminating unwanted noises in vehicle and system.
Find peak resonance of system.	Sweep vehicle with sine wave output and find frequency that system will have highest (most efficient) output.	Help select software that will produce the highest sound pressure level for the vehicle/system.
Adjust RTA (Real Time Adjustment).	Using pink noise output of OSC2, measure frequency response of system, make adjustment to the system and note the changes in response.	Easy method of measuring and adjusting the system for smooth octave to octave spectral balance.
Measure transfer function of vehicle	Plot frequency response of vehicle with test enclosure and subtract near-field response of test enclosure from measurement.	Inputting transfer function of the vehicle into Term-Pro will provide user with more accurate box response predictions.

Feature	Procedure	Benefit
Set gain controls of system.	Use Track 99 (all bits high) on CD to adjust system so that maximum input signal will not clip electronic components of system.	This process will provide maximum signal to noise ratio, minimum reliability of all components in system.
Determine tuning frequency of vented system.	Using Tracks 10-98, measure actual tuning frequency of enclosure and adjust to desired tuning frequency.	Ensures maximum performance from a vented system.
Find resonance of vehicle.	Using Tracks 4, 5, 10-98 to sweep the vehicle, find what frequencies cause rattles and buzzes then damp the unwanted noises.	Quick, easy method of locating and eliminating unwanted noises in vehicle and system.
Find peak resonance of system.	Using Tracks 10-98, sweep vehicle with sine wave output and find frequency that system will have highest (most efficient) output.	Help select software that will produce the highest sound pressure level for the vehicle/ system.
Determine absolute polarity of a system.	Using Track 3 and PD1, measure polarity of system from source unit to speaker.	Only method to determine absolute polarity for best sonic performance.
Determine F3 of a low frequency system.	Use Tracks 10-98 to set system level to medium output and locate the frequency where the system's output reduces drastically.	Use F3 to design and install a sub-sonic filter that will limit output and excursion below speakers operational bandwidth reducing distortion and improving output capability.
Determine sonic characteristics of low frequency system.	Using Track 9, listen to the attack of the low frequency system (a dull sounding attack indicates poor transient response) small improvements mounting tuning and adjustment are very noticeable.	Easy way to determine and maximize the transient performance of a low frequency system.

PD1/PD2

(Pulse Generator & Detector)

- **Determining Driver Location and Polarity**

When installing an audio system, two very important factors determine its sonic potential: ***Location*** and ***Polarity***.

Location is obvious. Selecting the proper location and wiring the driver in that location is a major factor in the system's ability to recreate a sound stage. **Polarity** is less understood. Most people can understand that if two speakers (a left channel and right channel midrange) are out of polarity, constructive/destructive interference can possibly degrade the performance of the system.

What is equally important is the absolute polarity of the system. Software (the music we listen to) was recorded with the understanding that the end user's system would reproduce the information as it was recorded. Some may argue that as long as the relative polarity between drivers is correct, the result will be the same. A simple experiment with the PD2 will demonstrate this is not the case.

To compensate for phase shift caused by passive crossovers, manufacturers invert the polarity of one of the drivers (usually the high frequency driver). The driver's terminals will be incorrectly marked to simplify the installation for the installer. If you decide to upgrade to a fully active system, the terminal markings would be incorrect. Relying on a manufacturer to properly mark drivers could result in a compromise that would adversely affect the performance of a system. Determining polarity of a driver can only be done by testing. Listed below are some examples in determining driver location and polarity.

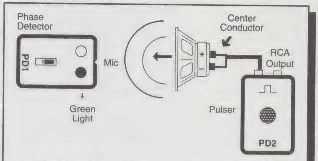
- **Test Battery and Visual Inspection**

Advantage:	Easy to use (only if driver is visible)
Disadvantage:	Harmful to drivers.
	(The smaller the driver, the greater the possibility of damage)
	Can only determine driver polarity if visible
	Difficult to determine absolute polarity of system

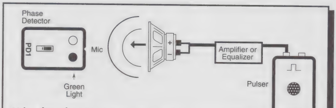
- **PD1/PD2 (with use of Test CD #101)**

Advantage:	Easy to use
	Safe for all drivers
	Senses driver pressure to determine polarity
	Determine absolute polarity of system when used with CD #101
Disadvantage:	Only available from Rockford
	(Disadvantage to other dealers)

PD2 output will drive either an amplifier/preamp signal or a speaker directly.

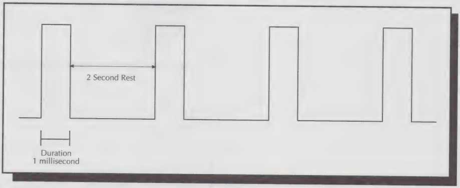


Using the Pulser to Directly Drive a Speaker with Positive Pulses



Using the Pulser to Supply Positive Pulses to an Amplifier or Equalizer for Polarity Checking.

Output from PD2 (from .1V to 4V @ 4Ω) is a **Square Wave Output**.



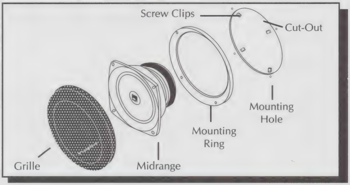
The **PD1 Phase Detector** is a small, handheld, battery operated diagnostic tool designed to quickly and accurately determine the relative phase of a speaker or a speaker system. There is an internal microphone in the PD1's front panel and two top-mounted LEDs (light emitting diodes) which indicate the following:

GREEN = Speaker cone is **pushing towards** the PD1's microphone or
RED = Speaker cone is **pulling away** from the PD1's microphone.

The only control on the device is the on/off rocker switch which should be kept in the off position when not in use.

• **BACKGROUND**

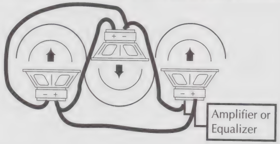
It was long ago determined that sound waves travel through the air at around 1130 feet per second. In the last 100 years we have been creating sound by forcing speakers to move back and forth in the air. (Speakers cannot work in the vacuum of space). The actual motion of a speaker's cone is the result of varying electrical currents flowing in the speaker's voice coil. This voice coil is typically constructed of a double winding of a small gauge solid enameled copper wire. A magnetic field is placed in and around the voice coil by way of a permanent magnet.



The direction in which the electrical current is flowing through the voice coil determines if the speaker's cone will move forward or backward. Conventionally, if a small direct current from a flashlight battery is made to flow through the voice coil and the speaker's cone moves forward, then the positive (+) terminal of the voice coil is the one connected to the positive (+) electrode of the battery.

If the direction of the current is changed by simply reversing the connections to the voice coil, then the **Polarity** of the speaker is reversed. Also, the **Phase** of the audio information produced by the speaker is changed by 180°.

If one or more speakers in a speaker system is "out-of-phase" with the rest of the system, then the entire system will suffer the consequences of **Cancellation**. Cancellation disrupts the air by smoothing out the sound waves. One out-of-phase speaker can effectively cancel the sound from several in-phase speakers. It is extremely important for the polarity of each voice coil in the sound system to be wired "in-phase" with all the other voice coils in the system.



Cancellation due to Inverse Speaker Polarity has the same Effect as Physically Reversing a Speaker. The Middle Speaker above is Out-Of-Phase with the Others.

• **COMPLICATIONS**

Most speaker manufacturers place a red mark or a (+) on the positive speaker terminal. Some speaker manufacturers place the mark on the negative terminal! Most audio components do not invert the signal as it passes through its circuits, but some components do invert the audio!

Additionally, crossovers have the ability to produce phase differences between their outputs. These differences can have serious consequences. Let's examine some typical crossovers and their input and output phase relationships.

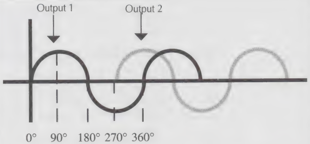
The **Order** of a filter is a measure of its rate of attenuation. An **Octave** is a doubling or halving in frequency. For instance, a 400Hz tone is one octave above a 200Hz tone, but a 100Hz tone is one octave below that same 200Hz tone.

Passive filters are series and/or parallel configured capacitors, inductors, and/or resistors constructed to function as a crossover. Active filters are electronic circuits utilizing op-amps, power supplies and sophisticated circuitry for crossover networks. Sharp cutoff rates and low-losses are features of active crossover networks.

A **Crossover** is a filtering device that either passes or rejects certain designated frequencies. There are two main qualities of a crossover: its order and its "**Q**". The "**Q**" of a filter is the sharpness of the filter's cutoff near the resonant frequency where the filtering action just begins to take effect.

A first order network, called a 6dB/octave crossover, will produce a 90° **Phase Shift** between its high and low outputs. This results in the delivery of both constant voltage and constant power through the crossover region. An advantage of the first order network is **All Pass**, which means that the sum of the output is identical to that of the input. Since first order networks have a very low rate of attenuation they can harm tweeters and mid-range speakers.

Second order crossovers, also called 12dB/octave networks, produce a 180° phase shift between their outputs. A typical application of a second order network would call for reversing the polarity of the higher frequency driver so that its ouput will be inverted. Now the tweeter will be in phase with the mid-range or woofer. Did the manufacturer of the 12dB/octave crossover already account for this polarity reversal when marking his device?



270° Phase Shift Between Outputs of an 18dB/octave Crossover

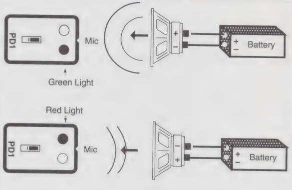
Fourth order crossovers, also called 24dB/octave networks, will keep both outputs in-phase, however, the roll-off associated with these crossovers is very sharp.

Higher order crossover networks follow the same pattern (i.e. 5th=450° difference, 6th=540°, 7th=630°, etc.). Note: Multiples of 360° can be subtracted from actual phase differences of higher order crossovers.

In general, it is very important to make sure that the phase of each and every speaker in a sound system is correct. Consideration for the phase shifts produced by the crossover network, as well as the speaker enclosure, must be taken into account when designing sound systems.

• OPERATION OF PD1

To use the PD1 to determine the polarity of a speaker, the speaker will first have to be driven, or pulsed, in a definite direction. For example, let's take a small flashlight battery and quickly connect the positive electrode to the + or red terminal of a test speaker. Connect the negative electrode to the - terminal of the same test speaker. The speaker cone should move out and forward when it is pulsed in this manner.



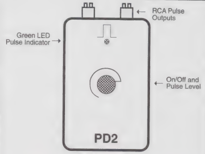
Verifying Polarity with the PD1 Phase Detector

When using the PD1 Phase Detector, try to keep all background noise to a minimum level. The microphone is very sensitive and will respond to outside audio interference.

Position the PD1 about one foot in front of the test speaker and make sure that the microphone is aimed directly at the test speaker. Using the battery as described above, quickly pulse the test speaker. The GREEN LED should momentarily light indicating the speaker was positively pulsed.

Now reverse the battery's leads and negatively pulse the speaker. The RED LED should momentarily light indicating the negative pulse was detected by the PD1. For demonstration purposes, a grille cloth can be draped over the test speaker illustrating the fact that the PD1 can "see through grilles."

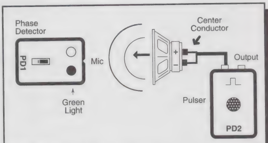
Using a battery to drive a speaker can harm the speaker by causing excessive direct current to flow through the voice coil. Once the voice coil is deformed, the speaker is essentially ruined. A better method of pulsing a speaker would be to use the PD2 Pulser. This battery operated and portable device will provide perfect positive pulses to drive a speaker or an amplifier/equalizer.



PULSER

• Using the PD2 Pulser

The PD2 Pulser provides a dual mono audio, positive output pulse approximately .001 seconds in duration. The pulses are spaced about 2 seconds apart to allow the speaker to come to rest. The female RCA plugs on the front of the PD2 are output ports and can pulse a speaker when the sensitivity adjustment is at its maximum level (4V peak) into a 4 Ohm speaker. When the sensitivity knob is set someplace below its maximum level, then the PD2 can provide a pre-amp level positive pulse to drive an amplifier, electronic crossover, equalizer, etc.



Using the Pulser to Directly Drive a Speaker with Positive Pulses

To demonstrate the function of the PD2 Pulser when used as a positive pulse source for the PD1 Phase Detector, position the phase detector about one foot in front of a test speaker. Connect the output of the PD2 Pulser to the test speaker's terminals and turn the sensitivity control clockwise to the maximum level. The center conductor of the RCA plug is the positive output and directly corresponds to the positive electrode of the battery. When wired so that the inner conductor of the RCA is connected to the + or red terminal of the speaker, then the speaker should pulse **TOWARDS** the phase detector. Sharp clicks should now be heard in the test speaker. Turn on the PD1 Phase Detector and the GREEN light should be pulsing with the speaker clicks.

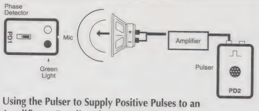
Now reverse the leads on the terminals. The **RED** light should illuminate, indicating that the speaker is pulling away from the microphone of the PD1. The PD2 Pulser can be used in this manner to safely provide positive pulses to sub-woofers, woofers, mid-range speakers and tweeters.

• Preamp Level Pulsing

To use the PD2 Pulser to drive through an amplifier, equalizer, or electronic crossover, simply connect the PD2 to the input of the component to be pulsed with male-to-male RCA cables. Although the PD2 has two bridged female RCA plug outputs, it may be desirable to use a single RCA cable and move it between the various amplifier channels in the sound system.

After the RCA cable(s) are connected between the pulser and the component, activate the amp/eq/xover and then slowly turn the knob clockwise until the pulses can just be detected in the speakers. Next, turn on the PD1 Phase Detector and, with its microphone aimed directly at each speaker to be tested, check the polarity of each speaker.

If many speakers are installed in a vehicle, make sure to check only one speaker at a time by placing the PD1 directly in front of that speaker. To ensure accurate measurements, disconnect all speakers except the one speaker to be tested.



Using the Pulser to Supply Positive Pulses to an Amplifier or Equalizer for Polarity Checking.

Take care when pulsing tweeters with the PD2. There is no need to overdrive any speaker. The phase detector is very sensitive and if the pulse is audible, the PD1 will determine its polarity.

THE PD-TAPE is a stereo (two channel) audio cassette tape that was individually mastered to provide positive pulses at the output of a tape player– even through the tape equalization. The PD-TAPE was recorded for approximately three minutes on each side.

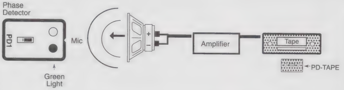
• Other Pulsing Methods

To use the PD-TAPE Positive Pulse Cassette Tape with the PD1 Phase Detector, simply turn on the sound system, insert the tape into the cassette mechanism, and adjust the volume of the system until the positive clicks can just be heard in the speakers. Now, position the PD1 directly in front of each speaker and check the polarity.

NOTE: When using the PD-TAPE in Auto-Reverse tape decks, the pulses will only be positive in one direction of tape travel. When the tape reverses, the polarity of the pulses will also reverse.

This means that auto-reverse decks invert the audio in one direction of travel. The test tape will produce all GREEN lights on one direction of travel, but as soon as the tape reverses, only RED lights should blink on the PD1.

The PD-TAPE can be used as a portable pulse source when used in a portable cassette player such as a Walkman-type device fitted with RCA connectors. The battery-operated portable cassette player can then be moved from car to car and connected in a manner similar to the PD2 Pulser.



Using the PD-TAPE Positive Pulse Cassette Tape

At the time of this writing, positive pulses are being mastered onto a demonstration CD (compact disc). The use of the CD demo disc is similar to that of the test tape, except that the CD will only produce positive pulses.

IM-1
(Impedance Meter)

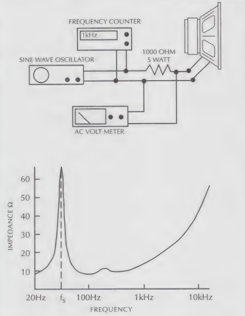
Measuring Impedance

- **Digital Voltmeter (DVM)**

Advantage: Easy to use
Disadvantage: Only measures DCR (Direct Current Resistance)

- **Voltage Divider Method** *(See Figure 10.1)*

Advantage: Accurate measurement of impedance versus frequency
Disadvantage: Expensive, user has to purchase multiple components
Difficult to accomplish testing in a vehicle



Note: Rise due to inductance of voice coil.

Figure 10.1

- **IM-1**

Advantage: Accurate measurement of impedance versus frequency
Portable, unit is self-powered
Easy to operate
Disadvantage: Only available from ROCKFORD (Disadvantage to other dealers)

Understanding Impedance

Impede, the root word of impedance, means to resist against something. Many people mistake this information to say that impedance is just resistance and can be measured with a multimeter. Impedance is much more complicated than this. Impedance is not a static, but a dynamic measurement. This means that the impedance of a speaker or speaker system varies with frequency and other parameters. In simple terms, impedance is an AC resistance measurement which includes three separate factors: DC resistance, Capacitive reactance and Inductive reactance. The sum of all these components determine the actual impedance. The impedance can be measured with an impedance meter such as the IM-1.

- **Resistance & Reactance**

The three parameters that determine the impedance are the DC resistance, the Capacitance reactance and the Inductive reactance. The **DC Resistance** is the real component of the impedance. Real component means it can be directly measured. The DC resistance is the resistance of the wire in the voice coil. This is commonly referred to as the DCR of the speaker. It can be measured with a multimeter set to the "Ω" position. Many people commonly believe that this is the impedance of the speaker. This is incorrect. The DC resistance is only **ONE** of the parameters that determine the impedance. The **Capacitive reactance** (X_C) and **Inductive reactance** (X_L) components of impedance are the imaginary components. Imaginary means these components cannot be measured directly and have to be extrapolated to determine their value. To best understand how all three components combine to form the impedance, let's look at the vector analysis of impedance.

The vector analysis of impedance may overwhelm some people. To better understand impedance, let us use an analogy of a man (or woman) walking in a park. He is sitting on the park bench. Let us refer to the bench as the reference. Directly in front of him is the snack bar. Directly to the left is the river. Directly to the right is the parking lot where his car is located. First he takes 4 steps towards the snack bar. He then turns left and takes 1 step towards the river. He hears his car alarm go off and turns 180° around and takes 4 steps toward the parking lot to see if someone is trying to steal his stereo. How far is he away from the park bench?

By using vector analysis we can easily solve this problem. As support material we will need two pieces of graph paper. On one piece of paper we will draw our vector model. The other piece of paper we fold into a ruler. Draw an X and Y axis on the graph paper (two lines that are perpendicular to each other.) *Figure 10.2.* Where the lines intersect is your starting point or the park bench.

IM-1
Impedance Meter
Operation Manual

The IM-1 is designed to test the impedance of speakers, passive crossover networks and speaker systems over the full audio frequency range.

• Specifications

- Maximum Impedance Measurement: 100Ω
- Impedance Measurement Accuracy: 0Ω - 50Ω: 5%
  50Ω - 100Ω: 10%
- Frequency Range: 20Hz-20kHz in 3 ranges
- Frequency Measurement Accuracy: 1%
- Typical Open Circuit Voltage At Probe Tips: 1.5 Vrms

• Features

- Hand Held Operation
- Operates on 9V Battery
- Impedance Measurement Range: Full Audio Spectrum/20Hz-20kHz
- Two Operating Modes:
  - A. Frequency Dependent Impedance Measurements
  - B. Frequency Counter
- Built-In Signal Generator
- Low Battery Indicator
- Ideal for Determining Circuit Impedance & Frequency of Measurement
- Perfect for Troubleshooting and Characterizing Systems
- Protective Carrying Case

• Warnings

1. **NEVER** connect or disconnect any component of the sound system with power applied to the system.

• Low Battery Indication and Battery Replacement

When the battery voltage drops too low for proper operation, the IM-1 will read "0.0" at all times in the impedance mode. The open circuit reading of the IM-1 indicates the condition of the battery during operation. The open circuit reading decreases as the battery voltage falls. When the open circuit reading decreases to about "120.0", the battery voltage becomes too low and the display reading changes to "0.0". An IM-1 with a low battery will read "0.0" in the impedance mode at all times no matter what the actual circuit impedance is. For maximum battery life, use an alkaline battery to power the IM-1.

When replacing the battery in the IM-1, use the following precautions:

1. Remove the IM-1 test leads from the circuit under test.
2. Turn off power switch on the IM-1.
3. Install battery correctly into battery connector. An incorrectly installed battery will become hot and will not allow the IM-1 to operate.

Failure to follow the battery replacement precautions may result in damage to the battery or tester.

• **APPLICATIONS**

The IM-1 measures the impedance of a circuit by supplying an A.C. current to the circuit under test and measuring the voltage developed across the circuit. Since the test signal from the IM-1 is A.C., the IM-1 can measure circuit impedance measurements, detection of open or short circuits in speaker systems, and frequency response characterization of speakers and speaker systems.

• **TESTING PROCEDURE**

To use the IM-1, the following procedural steps are necessary to ensure accurate measurements.

STEP 1 Move the **FUNCTION** switch to the **IMPEDANCE** position.

STEP 2 Use the **FREQUENCY RANGE** switch to select the desired range for testing. Options are: 20Hz - 200Hz, 200Hz - 2kHz, 2kHz - 20kHz.

STEP 3 Turn the IM-1 to the ON position by rotating the **FREQUENCY ADJUSTMENT** knob clockwise until you feel a slight "click."

STEP 4 Plug the test leads into the IM-1. Observe polarity (red +, black –).

With the test leads open circuited (not connected to anything), the LCD display on the IM-1 will read from "120.0" to "135.0" depending on the state of charge of the internal battery.

If the display reads "0.0" check the internal battery for proper connection and state of charge. If the battery is bad replace it with a new one.

WARNING: Batteries that are in a discharged state can leak acid which can damage the IM-1. Discharged batteries should be removed immediately. If the IM-1 is to be stored for a period of more than 30 days, the battery should be removed.

STEP 1 To test a speaker system/circuit use the following procedures:

STEP 2 With the IM-1 set up as described by the above procedures attach the test leads to the speaker wire that "feeds" the speaker system. The wire must be disconnected from the amplifier. It is best to check one channel at a time. i.e. Test left front then right front and so forth. Be sure to observe polarity.

STEP 3 Observe the LCD readout. A nominal impedance will be shown.
Example: If the speaker wire to which the IM-1 is attached feeds a speaker system/circuit that includes a 4 Ohm woofer, a 4 Ohm midrange and a 4 Ohm tweeter all wired in parallel with assorted passive crossover components, then the IM-1 readout should be approximately "3.5". This is with the **FREQUENCY RANGE** switch set in the 20Hz - 200Hz position and the **FREQUENCY ADJUSTMENT** fully counterclockwise.
To observe the frequency at which the impedance reading is being taken set the function switch to the **FREQUENCY** position. In the previous example, the readout should read approximately "20" when the function switch is moved to the **FREQUENCY** position.

STEP 4 By rotating the **FREQUENCY ADJUSTMENT** switch clockwise it is possible to vary the frequency within a selected range. Moving clockwise increases the frequency and counterclockwise decreases the frequency.

This allows for impedance measurements to be taken across the entire audio spectrum. With the function switch in the **IMPEDANCE** position, slowly rotate the **FREQUENCY ADJUSTMENT** knob clockwise while observing the display readout. If the impedance measurement drops below the amplifiers' "safe operating range", switch the function switch to the **FREQUENCY** position to isolate where the problem exists. i.e.: A low impedance measurement at 300Hz could indicate a poor crossover design, a shorted capacitor or induct coil, overlapping crossover frequencies, etc.

OSC2

Sine Wave/Pink Noise Generator

Verify Tuned Frequency

• Why is tuning Frequency (F_b) Important

Once you have constructed your bass reflex enclosure, it is critical to verify the tuning frequency of the enclosure. Any error in the tuning frequency can have disastrous results depending on the design of the enclosure. In High "Q" Sub-Sonic Filter Applications, small deviations in the tuning frequency will create a situation where the maximum excursion of the speaker can easily be exceeded resulting in a loss of reliability. This would be counterproductive to the filter's design goals. Relying only on calculations will not guarantee proper performance.

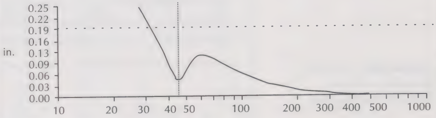
There are several methods possible to measure the tuning frequency. The IM-1 can be used to determine the impedance plot of a speaker in an enclosure and the tuning frequency can be derived from that measurement. An easier form of determining the tuning frequency of the enclosure is to use an OSC2 and use one of the following procedures.

• Tools Needed

- OSC2
- Power Amplifier (any size)
- 50Ω 20 Watt resistor
- Voltmeter
- Source Unit (Optional)
- Test CD #101 (Optional)

• Visual Method

The visual method consists of sweeping the enclosure with a sine wave and observing the cone movement of the speaker. In the proximity of the tuning frequency, the cone movement will appear to lessen to a point where the speaker's movement appears to stop. This is the tuning frequency of the enclosure. An example of the cone displacement versus frequency of the speaker can be seen in *Figure 9.2*.



- At the F_b the cone motion is minimal
- Above the F_b , cone motion peaks typically less than one octave above the F_b of the enclosure.
- Below the F_b , cone motion of the speaker increases drastically as the frequency decreases.

Figure 9.2

STEP 1

Wire test gear to enclosure. Refer to *Figure 9.3* for wiring instructions. ***Note: Make sure all passive components except the test resistor are removed from the circuit.***

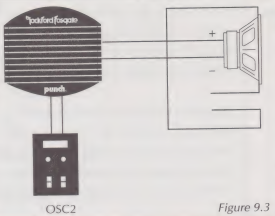


Figure 9.3

STEP 2

Sweep enclosure with generated sine wave. Using the OSC2 in the sine wave operation, sweep the enclosure with the OSC2 around the estimated tuning frequency with moderate output. (About 90-100dB SPL output from the speaker)

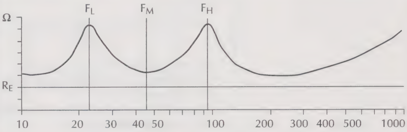
STEP 3

Determine minimum cone movement. While playing a sine wave into the speaker, gradually adjust the frequency output of the OSC2 until the cone motion of the speaker appears to stop. This is your tuning frequency.

• IMPEDANCE PLOT METHOD FIND FM

The impedance plot method differs slightly from the visual method with the addition of a resistor in series with the low frequency system (speaker and enclosure). In a standard bass reflex enclosure, the tuning frequency of the enclosure can be derived by measuring for F_M or calculating F_b from F_H and F_L. *(Figure 9.4)*

• IMPEDANCE PLOT OF WOOFER IN BASS REFLEX ENCLOSURE



- R_e is the DC resistance of the speaker
- F_L is the lower impedance peak of the speaker in the enclosure
- F_H is the higher impedance peak of the speaker in the enclosure
- F_M is the minimum impedance of hte speaker between the two impedance peaks F_L and F_H

Figure 9.4

• **Not the Radio FM**

The impedance peaks (F_L and F_H) can be located by using a voltmeter to read the output voltage from an amplifier across a resistor in series with the speaker in the enclosure. The impedance peaks occur when the voltmeter reads a minimum voltage. The impedance low (F_M) between the impedance peaks occurs when the voltmeter reads a voltage peak. Most of the time the F_M will be the tuning frequency of the enclosure. Although it is sometimes possible, that due to large inductance voice coils, the F_M will not accurately track the tuning frequency of the enclosure. In situations where the visual method and the Find (F_b) method results differ, another, more accurate method is available. For more detailed information on this other measurement, refer to the *Calculate F_b* section of this chapter. Listed below are the steps to determine the F_b of the enclosure by measuring the F_M.

Step 1 **Wire test gear to enclosure.** The resistor should be a 20Ω, 10 Watt resistor or other close value. Refer to *Figure 9.5* for wiring instructions. **Note: Make sure all passive components except the test resistor are removed from the circuit.**

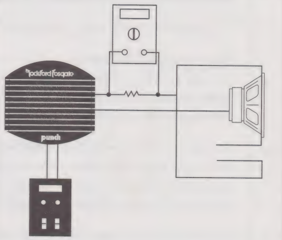


Figure 9.5

Step 2 **Sweep enclosure with generated sine wave.** Using the OSC2 in the sine wave operation, sweep the enclosure with the OSC2 around the estimated tuning frequency with moderate output. The input voltage into the enclosure should be around 5-10 volts.

Step 3 **Determine F_L and F_H.** Using an AC voltmeter, sweep the enclosure and find the frequencies where the minimum voltage occurs. These points surround the estimated F_b of the enclosure.

Step 4 **Determine F_M.** Using the voltmeter, sweep the enclosure with the OSC2 between the F_L and F_H and find the frequency where the maximum voltage occurs. This should be the tuning frequency (F_b) of the enclosure.

• CALCULATE F_b

The steps required to calculate the tuning frequency of the enclosure does not differ greatly from the steps used in *Impedance Plot Method.* The derived measurement increases the accuracy of the F_b of the enclosure by incorporating the F_c or the resonant frequency of the enclosure volume if the enclosure was sealed. The variables F_c, F_L and F_M combine in the following formula to determine F_b. *(Figure 9.6)*

$$F_b = \sqrt{F_L^2 + F_H^2 - F_c^2}$$

Where:

Fb = Tuning frequency of the enclosure
FL = Lower impedance peak in a bass reflex enclosure
FH = Higher impedance peak in a bass reflex enclosure
Fc = Resonant frequency of a sealed enclosure equal to the volume of the bass reflex enclosure

Figure 9.6

The steps to calculate the F_b are as follows:

STEP 1 **Wire test gear to enclosure.** The resistor should be a 20Ω, 10 Watt resistor or other close value. Refer to *Figure 9.1* for wiring instructions. ***Note: Make sure all passive components except the test resistor are removed from the circuit.***

STEP 2 **Sweep enclosure with generated sine wave.** Using the OSC2 in the sine wave operation, sweep the enclosure with the OSC2 around the estimated tuning frequency with moderate output. The input voltage into the enclosure should be around 5-10 volts.

STEP 3 **Determine F_L and F_H.** Using an AC voltmeter, sweep the enclosure and find the frequencies where the minimum voltage occurs. These points surround the estimated F_b of the enclosure.

STEP 4 **Seal the port on the enclosure.** For accurate results be sure to make an air tight seal on the port.

STEP 5 **Determine F_c.** Using the voltmeter, sweep the sealed enclosure with the OSC2 between and determine the frequency with the minimum voltage (maximum impedance peak). This is the resonant frequency (F_c) of the enclosure.

STEP 6 **Use the following formula and calculate the tuning frequency F_b.**

$$F_b = \sqrt{F_L^2 + F_H^2 - F_c^2}$$

• CALCULATE F_b EXAMPLE:

The woofer used is an RFA-412. The enclosure used is a bass reflex enclosure with 2.35 feet of air volume in the enclosure. Both the port and the woofer are positioned on the same face of the enclosure. The impedance calculation was performed with an OSC2, a CD source unit with the Autosound Test CD #101 and voltmeter and verified with LMS (Loudspeaker Measurement System) data analysis program. The results are listed below in *Table 9.1.*

	LMS Data	OSC-2/CD #101 Data
F_L	15.05Hz	15Hz*
F_H	53.79Hz	54Hz
F_c	41.2Hz	41Hz

Table 9.1.

***Note: Some OSC2s are not capable of reproducing frequencies below 20Hz. When measurements below 20Hz are required, use a CD source unit and the Test CD #101 to complete the test data.**

• CALCULATIONS WITH LMS DATA

$$F_b = \sqrt{F_L^2 + F_H^2 - F_c^2} \Rightarrow$$

$$\sqrt{15.05^2 + 53.79^2 - 41.3^2} \Rightarrow$$

$$\sqrt{226.5025 + 2893.3641 - 1705.69} \Rightarrow$$

$$\sqrt{1414.1755} \Rightarrow 37.605539 \approx 38Hz$$

• **Calculations with OSC-2 Data**

$$F_b = \sqrt{F_L^2 + F_H^2 - F_c^2} \Rightarrow$$

$$\sqrt{15^2 + 54^2 - 41^2} \Rightarrow$$

$$\sqrt{225 + 2916 - 16.81} \Rightarrow$$

$$\sqrt{1460} \Rightarrow 38.209946 \approx 38Hz$$

Volume of an Irregular Shape

When designing a low frequency system, an enclosure is usually required to accomplish performance goals. It is not always possible to construct an enclosure that resembles a basic shape. Often, one is required to build an enclosure that forms to the interior of the vehicle, like the spare tire well. Calculating the air volume of this location can prove to be difficult.

In the past, installers would use several methods to calculate the air volume. One method would be to divide the enclosure into separate basic shapes. In irregular types of enclosures this will only give an approximation and is not entirely accurate.

A second method would be to fill the enclosure with a material and then pour this material into a rectangular shaped enclosure and measure the volume the material displaces. While this method is more accurate than the first, the accuracy depends of the compressibility of the material used.

Another method not commonly used is the most accurate method of all. It involves measuring the volume using the OSC2 Sine Wave Generator. Using the formula below, the volume of the enclosure can be calculated by determining the port tuning frequency of a know port length.

• **FORMULA FOR CALCULATING AIR VOLUME OF AN IRREGULAR SHAPE**

$$V_b = \frac{8466r^2}{f_b^2\,(L_v + 1.463r)}$$

Where:

V_b = Air volume of the enclosure in cubic feet

r = Radius of the port in inches

f_b = Tuning frequency of the enclosure with the given port in Hertz

L_v = Length of the port material used in inches

TOOLS NEEDED

- OSC2
- Power Amplifier (any size)
- Voltmeter (optional)
- Test Port Material (2"-3" material works)
- Test Plate for the speaker and port
- Test Speaker (6"-8" woofer works)
- 50Ω 20 Watt resistor (optional)

STEP 1 **Construct the enclosure.** While this appears to be the most obvious, obvious things are often overlooked.

STEP 2 **Build test plate for both the test speaker and the test port.** *(Figure 9.7)* It is beneficial to construct two different sized plates to cover enclosures designed to use 8" speakers up to 18" speakers.

STEP 3 **Place test plate on enclosure.** Make sure this is an air tight seal. If the test plate leaks, the measurement will be greatly affected.

STEP 4 **Sweep the enclosure with the OSC2 and determine the f_b or tuning frequency of the enclosure.** Refer to *Verify Tuned Frequency* in the OSC2 section in this manual for detailed instructions. ***Note: Make sure all passive components except the test resistor are removed from the circuit.***

STEP 5 **Input data into formula for calculating volume.** This will give you the actual volume of the enclosure to be used for port calculations or woofer and enclosure type selections.

The test plate is mounted to the enclosure. Using the Visual Method of determining the tuning frequency of the test woofer, we determine the tuning frequency of the enclosure. The resulting data is listed below.

Example:

Data: r = 2 inches

f_b = 34Hz

L_v = 10 inches

• **Calculations**

$$V_b = \frac{8466r^2}{f_b^2\,(L_v + 1.463r)} \Rightarrow \frac{8466 \times 2^2}{34Hz^2\,(10 + 1.463 \times 2)} \Rightarrow$$

$$\frac{8466 \times 4}{1156\,(10 + 2.296)} \Rightarrow \frac{33864}{1156\,(12.296)} \Rightarrow$$

$$\frac{33864}{14214.176} \Rightarrow 2.3824103 \approx 2.38$$

OSC2 Sine Wave / Pink Noise Oscillator Operation Manual

The OSC2 Digital Sine Wave / Pink Noise Oscillator is designed to provide a convenient means for tweaking almost every autosound installation.

• **Specifications**

Output Frequency:	Continuously variable from 25Hz-20kHz
Output Amplitude:	Continuously variable; 700 mV RMS Max
Distortion:	0.15%

• **Features**

- Switchable Sinewave/Pink Noise Outputs
- Continuously Variable Sinewave Output in Three Frequency Ranges
- Adjustable Output Level
- Low Distortion
- Portable Battery Powered Operation
- Accurate Digital Readout

• **Warnings**

- To prevent possible system damage:
- NEVER connect or disconnect any component of the sound system while power is applied to the system.
- ALWAYS turn the OSC2 on BEFORE applying power to the system.
- ALWAYS turn the system off BEFORE turning the OSC2 off.
- ALWAYS verify that the volume control on the OSC2 is fully counter-clockwise (minimum) BEFORE turning the system on.

• **Connection to the System**

Connecting the OSC2 to an autosound system is as easy as plugging in 2 RCA plugs. Since the OSC2 is battery powered, there are no power wires or ground loops to contend with. The OSC2 can inject a signal anywhere line level signals are present in the system. Just unplug the existing source unit and plug in the OSC2.

• **Applications Testing for Proper Phase**

Set the output for sinewave and adjust the frequency between 20Hz and 60Hz. Slowly bring the volume up until a moderate output is obtained. Try reversing the polarity of one of the woofers. If the sound output increases, the drivers are in phase. If the output decreases, the drivers were in phase before and the polarity should be returned to its original configuration.

• Setting Crossover Points

Crossover points and midrange/tweeter polarity are easily checked with the use of the OSC2's Pink Noise outputs. As with equalization, an RTA will be necessary. Overlapping crossover points will show up as Peaks in the response curve, while underlapping crossover points will show up as Dips. Since phase shifts occur in crossover networks, it is important that midrange and tweeters that are mounted in close proximity to each other be in phase. Otherwise, Hot Spots and Holes will occur in the frequency response of the system. Try reversing the phase of either the midrange or tweeter and see if the response on the RTA smooths out at the crossover point **THIS SHOULD BE DONE PRIOR TO EQUALIZATION.**

• Adjusting Equalization

By using the OSC2's Pink Noise outputs and a Real Time Spectrum Analyzer (RTA) an autosound system can be fine-tuned for an extremely smooth response curve. Connect the OSC2 ahead of the Equalizer. Slowly bring the volume up until a moderate output is obtained. Monitor the RTA display and adjust the equalizer for the smoothest response.

• Determining Resonant Frequency of the Vehicle

For people involved in Car Audio competition, the OSC2 can provide a substantial advantage. In order to obtain maximum Sound Pressure Level (SPL) scores at contests, it is important to know the resonant frequency of the vehicle. Set the OSC2 to sinewave and adjust the frequency to around 50Hz. Slowly bring the volume up until a moderate output is obtained. Now, slowly vary the frequency between 30Hz and 100Hz. At one point, the output of the system will increase. Observe the frequency displayed on the OSC2's digital readout. This is the resonant frequency of your vehicle. For competition purposes, select a song that contains a sustained note that very closely approximates this frequency.

• Setting System Levels

In biamped systems, the OSC2's Pink Noise generator is useful for setting the input levels on the amplifiers. Using an RTA, adjust the input levels on the amps for the smoothest response. **THIS SHOULD BE DONE PRIOR TO EQUALIZATION.**

• Troubleshooting an Installation with the OSC2

- Verify Operation of Source Unit
- Locate Ground Loops
- Locate Faulty RCA Cables
- Locate Faulty Line Level Accessories

• Tuning Vented Enclosures

The OSC2 is perfect for the installer who designs and builds vented enclosures. The variable output frequency makes it simple to determine the tuning frequency of the enclosure. Slowly bring the volume up until a moderate output is obtained. Now, slowly vary the frequency from 20Hz to 100Hz. At the tuning frequnecy, MINIMUM cone excursion will occur.

Low Frequency Test CD #101

Track One: This track contains a 1kHz tone recorded at a level of –20dB relative to digital clipping. This level is normally used in digital systems to indicate "0" level on normal program material. The level of this tone provides for 20dB of headroom and is an absolute level useful for comparison with other measuring systems. This track can be used for measuring the reference loudness of a component. This is accomplished by playing Track 1 on a CD player, and then measuring the signal output voltage of the headpiece with its volume control set to maximum.

Track Two: This track begins with a 1kHz tone, recorded at –20dB relative to digital clipping. After an initial period at –20dB, the level is decreased. Eventually, the level is reduced to digital zero, which continues for the remainder of the track.

This track is excellent for observing the reproducing quality of a sound system when driven with extremely low level signals. For these tests, the volume control of the sound system should be increased to keep the perceived audio level constant. The quality of the tone should not change and become distorted or fuzzy. This is a great track in which to test the sensitive circuitry of electronic noise gates. Track 2 is a repeatable source of low level signals for initial set up and calibration of noise gates. There is no need for special equipment.

For reference, the approximate levels referenced to track time are, :01 seconds = 20dB below clipping, :20 seconds = 30dB below clipping, :27 seconds = 40dB below clipping, :35 seconds = 50dB below clipping, :47 seconds = 60dB below clipping, :51 seconds = 70dB below clipping, 1:03 seconds = 80dB below clipping, 1:08 seconds = 90dB below clipping, 1:10 seconds = limited by CD player.

CAUTION: Plenty of time has been left on the end of this track to enable one to reduce the level of the volume control. **DECREASE THE VOLUME CONTROL NOW.**

Track Three: Both relative, as well as absolute polarity, can be quickly and easily measured using the high-frequency pulses contained on this track. The software is compatible and designed to be used with the PD-1 Phase Detector. In order to confirm accurate readings, every fourth pulse has been inverted. Readings can only be considered valid if the checker actually indicates this reversal when making the reading.

Relative polarity must be achieved if optimum performance is expected from a sound system. A single "out-of-phase" driver can wreak disastrous consequences in an otherwise great system. All the drivers in a system must be configured so as to work together. When Track 3 is played through a sound system, and the PD-1 is positioned directly in front of every driver, identical Phase Detector responses on each driver would indicate that the system is in relative polarity.

Absolute polarity is important in a system if accurate reproduction is the goal. Absolute polarity means that the sound system is reproducing the software exactly as it was recorded. Many musical instruments and voices have non-symmetrical waveforms and must be reproduced accurately with regard to polarity. Uniform polarity within a system is also important if the harmonics are going to be reproduced uniformly with respect to their fundamentals. The three positive pulses and one negative pulse that is repeated many times on Track 3 must be detected as such for a system to be in absolute polarity. ***Note: On the PD-1, a green light indicates that the speaker is pushing towards the microphone at the head of the device. A red light indicates that the speaker is pulling away from the microphone.***

Please take caution because the peak energy of this track is extremely high and could cause damage to sensitive tweeters. Keep the volume control at a modest level. The first minute of this track contains a low frequency background hum. With a normal full-range system, the level should be raised to the point where this hum is just barely audible. Take the readings very close to the individual speakers. The energy content covers the entire frequency spectrum and will give ample readings for loudspeakers covering the range from 20Hz to 20kHz.

Track Four: This track contains a slow sweep that covers the entire audible range from 20Hz to 20kHz. It is useful for making response measurements in conjunction with other test equipment. This track can also be used to check a sound system for any resonant components that may buzz or rattle during any part of the sweep. When using this track, listen carefully to the speakers for any signs of cone breakup or voice coil rubbing during the sweep.

The approximate track time vs. frequency relationship for Track 4 is as follows: :05 -=- 20Hz, :10 = 50Hz, :15 = 85Hz, :20 = 150Hz, :25 = 250Hz, :35 = 800Hz, :40 = 1.4kHz, :45 = 2.4kHz, :50 = 4.2kHz, :55 = 7.8kHz, 1:00 = 13kHz, 1:05 = 20kHz.

Track Seven: On this track there is a compilation of 16 free-running oscillators combined to generate a composite tone. The tones are totally non-synchronous in nature and are therefore more usable for spectral measurements with swept spectrum analyzers. This track can also be used to determine the overload threshold of a system particularly when equalizers are used in their boost mode. It is very easy to hear audible clipping on Track 7.

The frequencies generated in this composite are: 30Hz, 50Hz, 100Hz, 250Hz, 400Hz, 500Hz, 700Hz, 1kHz, 2.5kHz, 5kHz, 7.5kHz, 10kHz, 12.5kHz, 15kHz, 18kHz, 20kHz.

Track Eight: A 50Hz square wave tone starts this track. After a short period of time, this square wave sweeps down to 20Hz. This is useful for setting up low frequency servo systems.

Track Nine: This track contains low frequency tone bursts from 40Hz to 200Hz, in 10Hz increments. When these tones are played, they should sound sharp and clear with no hangover. Track 9 is very useful when evaluating the transient behavior of a sub-woofer system. Any tendency for the system to dull the attack of the bursts is an indication of poor transient response. Small improvements can easily be heard with this test track. The period between each burst is chosen to allow adequate time for the sound to decay away and there should be at least some total silence between bursts. Lack of this silent period is an indication of poor damping or an uncontrolled resonance in the listening area.

For a more exacting test, take a microphone and feed it directly into an oscilloscope. The actual waveform is two complete cycles of each frequency starting and stopping at zero crossover. The waveform should completely stop before starting again. Because of the dimensions of the automobile, these low frequencies are going to represent a pressure response behavior and are not likely to be bothered by standing wave phenomenon.

Tracks 10 through 98: With these tracks, the numerical track indicator of the CD player directly correlates with the frequency of the sine wave that is digitally recorded on that particular track. Track 10 is therefore 10Hz, Track 11 is 11Hz...Track 98 is 98Hz. There are no lapses between tracks and no breaks in the sine wave. Each track is approximately 30 seconds in length.

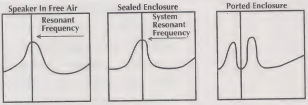
The most obvious use of these tracks would have to be determining the resonant frequency of a low frequency speaker and/or enclosure.

• **CONCERNING SPEAKERS ALONE**

Although there are many methods of performing the measurement, the concept is that at its resonant frequency a speaker is extremely efficient. Also, the speaker will reach its maximum impedance when at resonance. Consequently, when a speaker is in resonance, the current through that speaker's voice coil will drop to its lowest level. It follows then, that the voltage across the voice coil will reach its maximum when the speaker is in resonance.

• **CONCERNING SPEAKERS IN ENCLOSURES**

A speaker in a completely sealed box has characteristics similar to a speaker in free air and will exhibit a single peak voltage across its terminals when in resonance. However, a ported box will have two peaks and a dip between those peaks. The dip between the peaks correlates to the resonant (or tuning) frequency of the ported speaker system.



Examples of Voltage across Speaker Terminals (Vertical) vs. Frequency (Horizontal) as displayed on Track Indicator of CD.

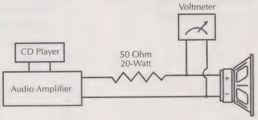
It is our suggestion that an analog AC voltmeter be used to measure the voltage across the voice coil of the speaker to be tested. Also, a 50Ω, 20-Watt power resistor should be inserted in series with + output of the amplifier and the + terminal of the speaker.

With the power resistor wired into place and the voltmeter connected across the terminals of the speaker, insert Test CD #101 into the CD player and set the volume loud enough to barely interfere with normal conversation. Adjust the scale on the voltmeter for a half-deflected needle.

Let's say that we are to determine the resonant frequency of a 10" woofer in a completely sealed box. Start with 50Hz (Track 50) and note which way the needle deflects as the track numbers are increased through Track 51, Track 52, etc. If the voltage drops as the track numbers are increased, then the resonant frequency must be lower.

Quickly scan through Track 49, Track 48, Track 47, etc. The voltage across the speaker terminals should now be increasing indicating that you are approaching the resonant frequency. Remember that for a speaker in a completely sealed enclosure you are looking for a maximum speaker voltage on the AC voltmeter. Keep changing CD tracks until theneedle stops peaking and then just begins falling back down. Some adjustment to the scale on the voltmeter may be necessary. Read the track number on the display of your CD player when the voltage is at its peak. The track number is also the resonant frequency, in Hz, of the speaker in its enclosure.

A digital multi-meter can also be used for these tests, but be aware that the low frequency response of most inexpensive DMMs leaves much to be desired. By the way, an easy way to verify the low frequency response of your DMM is to plug the signal output of the CD directly into the DMM. Using Track 4, note any variations as the signal sweeps between 20Hz to 20kHz. The AC voltage should not vary on a high quality meter. For a more detailed evaluation of your DMM, the low-frequency software on Tracks 10 through 98 can be used.



Test Arrangement for Determining Resonant Frequency of a Speaker, or System, using Low Frequency CD #101.

The value and wattage of the series resistor is also non-critical. The volume level of the CD player as well as the power output of the amplifier should not drastically change the results of this experiment. Also, if you prefer to measure the voltage across the series resistor (a minimum voltage would be present here at resonance), or the current flowing through the voice coil (minimum current would be present at resonance), the results should prove identical.

Track 99: Caution must be used with this track. It contains a 1kHz sine wave recorded at the highest possible level acceptable for a CD. The first and most obvious use of this track is to test the output headroom of a CD player.

The track begins at a relatively low level and gently rises to its maximum so as to provide adequate warning. At approximately 20 seconds, the waveform actually goes into clipping and then slowly reduces in output until at :27 seconds it remains just below clipping. Track 99 remains at this level until the end of the track.

Without the need for any other test equipment, Track 99 can produce an accurate test for clipping in a CD player. Listen carefully to this track with the volume turned way down and "learn" the sound of clipping as the time indicator advances from :20 seconds to :27 seconds. Next, increase the volume until you hear a similar sound or until the volume is turned all the way up. If your unit is clipping already, it will have a tendency to mask the portion that is intentionally clipped.This procedure can be used to optimize the gain structure of an entire system.

If an oscilloscope is available, much can be discovered about the quality of the Digital-to-Analog convertor and filters within a CD player during the intentionally clipped section. A really good design will show a minimum of ringing as well as a smooth flat top on the clipped sine wave instead of very ragged uneven clipping.

Chapter 10

Vehicle Acoustics

- Woofer Performance Characteristics Affected by the Vehicle
  - Measuring Vehicle Acoustical Transfer Function
    - V.A.T. Function, Term-Pro and Enclosure Applications

Vehicle Acoustics

Woofer Peformance Characteristics Affected By the Vehicle

• Acoustic Coupling

Acoustic coupling between multiple woofers occurs when the woofers are close enough to act as one piston. Locating woofers in the corners of the vehicle will not take advantage of this effect and the distance separating the woofers can cause destructive interference. In this case, destructive interference is the acoustic loss between the woofers at a particular frequency. This frequency can be determined by the formula in the *Standing Wave* section in this chapter.

• Boundary Effects

Woofers radiate low frequency acoustical output in a spherical pattern. This means the low frequency energy travels in all directions. This is true only for a free-field situation. A free-field situation exists when there is no interference with the acoustical output of the speaker. Any time a boundary is placed in the path of the woofer's acoustical output, the energy of the woofer output is reflected at an angle opposite to the angle of incident. The incident angle is the angle the acoustical output of the woofer strikes the boundary. The more boundaries placed against the output of the woofer, the greater the focus of the acoustical energy of the woofer in a particular direction.

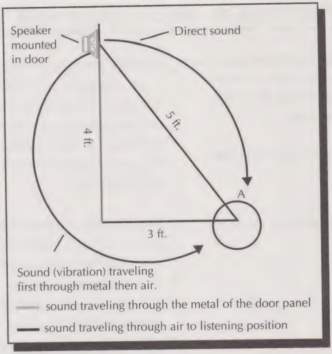
• Absorption and Reflection

When the acoustical output of the woofer or any speaker strikes a boundary, two possibilities occur: absorption or reflection. With absorption, the acoustical energy of the output of the speaker is absorbed and converted into vibrations and heat. The result is that the energy reflected off the boundary is greatly reduced. Reflecting is the redirection of the acoustical energy received by the boundary. A 100% reflective surface would reflect all of the acoustical output received. This never happens. All boundaries have a combination of both absorption and reflection.

• Transfer of Energy

One way to maximize your environment is to provide a good platform for your speaker mounting. When mounting the speaker on door panels, not only is it important to have a secure mounting location, but it's also important to prevent the vibration of the speaker from traveling down the door panel. What effect could this vibration cause? Below are the arrival times of direct sound of the speaker and sound (vibration) transmitted first down the metal panel.

Question:	Which sound will arrive first?
Givens:	Speed of sound in air (STP) - 1130 ft./sec.
	Speed of sound in metal - 16,600 ft./sec.



Medium of Sound	Speed of Sound Ft/Sec	Meters/sec.
Air (STP)	1,130	344
Sea Water	4,900	1,500
Wood (fir)	12,500	3,800
Steel (bar)	16,600	5,050
Gypsum Board	22,300	6,800

The sound traveling through the metal of the door panel reaches the listening position ahead of the radiated "direct" sound of the speaker. If an analogy of two speakers is used, the difference in time arrival (1.5ms) would equal a difference in distance of 1.695 feet between the two speakers relative to the listening position. Obviously, the resonance of the door panel will degrade sound quality. By damping the panel (with the Punch Mat or other damping material) the effect of this door resonance can be reduced at least 60dB* below the music level. Damping panel resonances will provide the platform necessary for awesome stereo reproduction.

* 60dB is equal to the human threshold of hearing. If a test tone A is 60dB than test tone B, then test tone A will be loud enough that it will mask or "wash out" test tone B. Mini-disk utilizes this knowledge of human hearing to compress the digital audio signal. By eliminating all musical information that is 60dB below other musical information, the saved space on the disk enables this format to have the same playing time as compact disc but be considerably smaller in size.

$$\boxed{\text{Direct Sound}} \; — \; \frac{5\,\cancel{\text{ft}}.}{1130\,\cancel{\text{ft}}/\text{sec}} = 0.0044 \text{ or } \boxed{4.4 \text{ ms}}$$

$$\boxed{\text{Door Panel}} \; — \; \frac{3\,\cancel{\text{ft}}.}{1130\,\cancel{\text{ft}}./\text{sec}} + \frac{4\,\cancel{\text{ft}}.}{16{,}600\,\cancel{\text{ft}}./\text{sec}.} = .00290 \text{ sec. or } \boxed{2.9\text{ms}}$$

• Standing Waves

Standing Waves are created by reflections between two parallel boundaries. The constructive and destructive quantity of the standing wave depends on the frequency of the standing wave, the distance separating the two boundaries relative to the frequency of the standing wave and the absorption characteristic of the boundaries. The acoustic output of the system is compromised of constructive (increase in output) and destructive (decrease in output) interference. The frequencies where potential standing waves can occur can be determined by dividing the number 565 the distance between two parallel boundaries in feet.

$$\boxed{f = \frac{565}{d}\, n_l \qquad n_l = 1, 2, 3, \ldots}$$

The result and multiple of the result are the frequencies where potential standing waves will occur. This formula can also be applied to determine the starting point of the cabin gain of the vehicle.

• Cabin Gain

Cabin Gain of the vehicle is the acoustical gain of a room or vehicle interior that is determined by the longest distance between to parallel boundaries. Below a certain frequency, the volume of air in the vehicle assists the output of the woofer. The increase in output averages around 12dB/octave from the starting frequency. The starting frequency can be determined by dividing the number 565 by the longest distance between two parallel boundaries or in the case of a vehicle the longest interior dimension.

$$\boxed{f = \frac{565}{d}\, n_l}$$

Below this frequency there is no longer the possibility of standing waves occurring in the vehicle. But this does not mean the woofer's output cannot be affected by another circumstance.

• Pressure Loading

The other circumstance is the potential pressure loading of the vehicle's interior on the woofer itself. Standard styles of enclosures can be greatly affected by this external force applied to the woofer. When the outside air pressure is changed, the ratio between the interior and exterior pressure exerted on the woofer changes. In sealed enclosures, this would change in the transient response of the woofer. This change would also change dynamically with amplitude. An analogy of this effect would be like setting up a race car suspension and then changing the weight of the car. The suspension is not set up for the different weight and could not react

properly. Race car tuners have a similar problem caused by the down force of the vehicle. As the speed increases, the added force of the wind pressing down on the car changes the effective weight of the car. The only way to maximize the performance is to take all these factors into consideration. If race car tuners don't take their environment into consideration, they lose the race – or worse. In car audio if you don't take the environment into consideration, the result will be poor sound or worse.

Interestingly, one of the benefits of the Aperiodic Membrane is the lack of reaction to the problem of pressure loading of the woofers. Because the resistance of the membrane is so much greater than the potential force applied by pressure loading, the ratio change between standard free-field pressure and the pressure loading in the vehicle will have little overall effect on the operation of the aperiodic system. What benefit does this have? The Aperiodic Membrane enclosure's inherent design maintains an excellent transient response not possible with other conventional enclosure designs.

Measuring Vehicle Acoustical Transfer Function

How can we compensate for these particular concerns? Well if we could measure the environment, we could design enclosures for that particular environment. We do this by measuring the VAT Function or the Vehicle Acoustical Transfer Function.

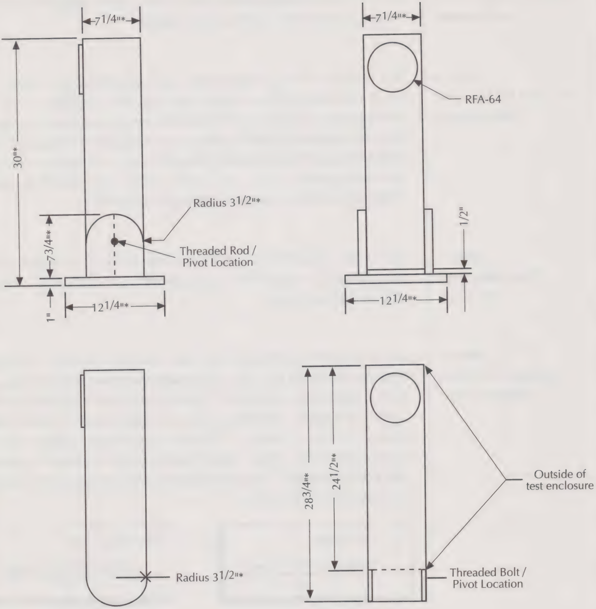
• Establish a Reference

First, we must establish a reference. Using a test woofer we can measure the free-field response. We then take that reference enclosure, place it in the vehicle and measure the response again. The difference between the transfer function of the vehicle.

Tools Needed

RTA Meter	AC Volt Meter
Test Enclosure	OSC2 Sine Wave Pink Noise Generator
Small Amplifier	CD Source Unit
CD #101	CD #102
Pencil	Work Sheet

TEST ENCLOSURE RFA-64



*Includes 1/8" Carpet thickness.
Total Volume approx. 0.48 cu. ft.
Enclosure wall thickness 1/2"

Transfer Function Data Worksheet Guideline

Step 1: Frequency in Hertz (Measured)

This is the test frequency output from your source unit. Frequencies from 10Hz to 98Hz are available from the Autosound 2000 Test CD #101. For frequencies above 98Hz, use the OSC2.

Step 2 Applied AC Voltage (Measured)

This is the AC voltage output from your amplifier. Use a digital volt meter and measure the voltage output of your amplifier at the test enclosure. Keep in mind not to exceed the maximum excursion capable of the test speaker. For best results, set the output level starting from the lowest frequency testing point, taking care not to exceed the maximum excursion capable of the test speaker. Select the desired frequency testing point while leaving the output at a fixed level.

Step 3 Measured SPL (In Car)

This is the measured SPL in the vehicle at the test location. This measurement is at the test frequency with the applied AC Voltage selected during testing.

Step 4 Converted Power (Calculated)

This is a conversion formula for data input into Term-Pro. Actually, Term-Pro calculates the SPL of a speaker/enclosure based off the AC Voltage applied to the speaker. Term-Pro relates this information (accepts and displays the information) as input power and/or power applied. To convert from AC Voltage over to Power (Watts), take the square of the input voltage and divide that by the nominal impedance rating of the speaker. This is the applied power used to determine the Calculated SPL in Term-Pro.

For a 4Ω Driver

$$P = \frac{(\text{Measured AC Voltage})^2}{4}$$

For an 8Ω Driver

$$P = \frac{(\text{Measured AC Voltage})^2}{8}$$

Step 5: Calculated SPL (Term-Pro)

This is the SPL predicted by Term-Pro in a free-field environment. The accuracy of this measurement will depend on the accuracy of the sensitivity rating by the manufacturer. Input the Converted Power (in Watts) from the Converted Power section into the SPL calculation of Term-Pro. Be sure the enclosure you use accurately tracks (resembles) the response of your test enclosure. For more information on selecting the Term-Pro test enclosure see the section *Build Your Reference Test Enclosure*.

Step 6: Transfer Function (ΔSPL)

To determine the Transfer Function, subtract the Calculated SPL (ΔSPL) from the Measured SPL (in car). The difference between the two is the Transfer Function (or Acoustical Gain) of the vehicle. This information can be imported into the *Overlay Design* section of Term-Pro for use when designing enclosures. If you do not have Term-Pro or another similar enclosure design program (shame on you). This information can be used as a reference material when choosing a box design of a speaker location.

TRANSFER FUNCTION DATA WORKSHEET

Vehicle: ______________________________ Date: ____________

Notes: ____

____

____

____

Frequency in Hertz	Applied AC Voltage (Measured	Measured SPL (In Car)	Converted Power (Calculated)	Calculated SPL (Term-Pro)	Transfer Function (Δ SPL)

TRANSFER FUNCTION DATA WORKSHEET

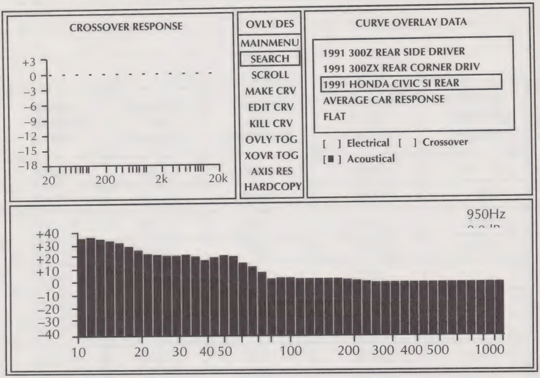
Vehicle: '91 Civic S1 Rear Date: 11/12/93

Notes: ____

Frequency in Hertz	Applied AC Voltage (Measured	Measured SPL (In Car)	Converted Power (Calculated)	Calculated SPL (Term-Pro)	Transfer Function (Δ SPL)
11Hz	3.53V	104dB	3.16W	69.5dB	+34.5dB
12Hz	3.60V	106dB	3.24W	71.1dB	+34.9dB
13Hz	3.68V	106dB	3.39W	72.4dB	+33.6dB
14Hz	3.74V	106dB	3.50W	73.6dB	+32.4dB
16Hz	3.79V	106dB	3.59W	75.3dB	+30.7dB
18Hz	3.83V	105dB	3.67W	77.4dB	+27.6dB
19Hz	3.79V	105dB	3.59W	80.3dB	+24.7dB
22Hz	3.79V	105dB	3.59W	82.0dB	+22.0dB
24Hz	3.62V	104dB	3.28W	82.8dB	+21.2dB
26Hz	3.82V	104dB	3.65W	84.5dB	+20.5dB
29Hz	3.80V	105dB	3.61W	84.5dB	+20.5dB
32Hz	3.79V	107dB	3.59W	85.3dB	+21.7dB
36Hz	3.72V	107dB	3.46W	87.0dB	+20.0dB
40Hz	3.70V	105dB	3.42W	87.8dB	+17.2dB
44Hz	3.63V	108dB	3.29W	88.6dB	+19.4dB
49Hz	3.60V	110dB	3.24W	89.5dB	+20.5dB
54Hz	3.55V	110dB	3.15W	90.3dB	+19.7dB
60Hz	3.48V	106dB	3.03W	91.1dB	+14.9dB
66Hz	3.45V	104dB	2.98W	92.0dB	+12.0dB
74Hz	3.27V	101dB	2.67W	92.8dB	+ 8.2dB
81Hz	3.36V	96dB	2.82W	92.8dB	+ 3.2dB
90Hz	3.34V	97dB	2.79W	93.6dB	+ 3.4dB

OVERLAY CURVE FROM TEST DATA

1991 Civic S1 - Spare Tire Well



CHAPTER 11

SETTING SYSTEM GAINS

- HECA SYSTEM FOR GAIN ADJUSTMENT
  - OUTPUT FROM CD101
    - ADJUSTING GAIN WITH BLT AND $250M^2$
      - ADJUSTING GAIN WITH BLT AND EPX_2

Setting System Gains

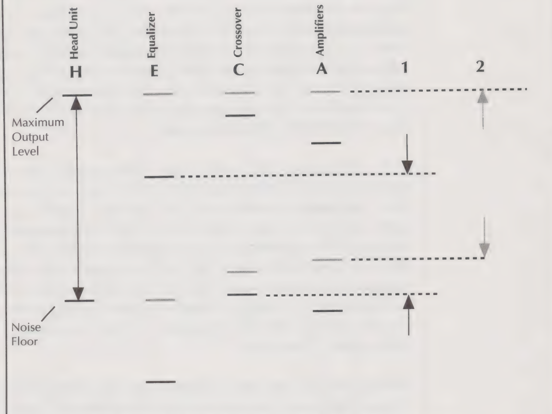
Once the system is up and running, the first step to maximizing the sonic potential and minimizing the possibility of system failure is to properly adjust the System Gain Structure. By adjusting the system gains, one is able to maximize the undistorted output while minimizing the noise floor. This relationship is referred to as the signal-to-noise ratio or the dynamic range of the system. Many people believe that system gains do not need to be set if you are using components from any one manufacturer. More than 99 times out of 100 this will not be the case.

In an audio system, the dynamic range is determined by the maximum undistorted output of the system relative to the noise floor. This noise floor can be interpreted in two different ways. The floor could be considered to be the ambient noise floor of the vehicle. This value is measured in dB by measuring the ambient noise of the sound system with the system's volume adjusted to the lowest position. In the IASCA and USAC forums, the ambient noise floor is determined using a recorded disk track of 0-bits. 0-bits is the technical term for no sonic information recorded with the system volume control adjusted to the highest position. Any noise emanating from the sound system is undesirable.

If the system gains are adjusted properly, the dynamic range of the entire system is determined by the signal-to-noise ratio of the worst component in the audio chain. If the system's gains are not adjusted properly, the dynamic range can be much worse to nonexistent. *Figure 11-1* illustrates a normal system set with factory settings. The top bar references the maximum output of the component and the bottom bar is the noise floor of each of the components. The space in between is the dynamic range of the component. The dynamic range of the system is determined by the lowest output and the highest noise floor.

The object of setting gains is to adjust all the audio components' input and output gain structure (whichever they have) so each component can pass the maximum signal that is undistorted. By doing this, each component's capabilities will be matched with the components that preceded and follow it in the audio chain.

HECA SYSTEM FOR GAIN ADJUSTMENT



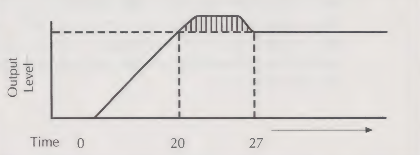
1 Dynamic Range of an unadjusted system (system left at factory gain settings)

2 Dynamic Range of system with gain levels

Figure 11.1

OUTPUT FROM CD101 (TRACK 99)

(0dB is the maximum undistorted output available from CD recordings)



From 20-27 seconds, track 99 is digitally clipped to give users a reference to what clipping looks/sounds like.

- **ADJUSTING GAIN WITH BLT AND 250M²**

STEP 1 **Install and wire source unit, BLT and 250m².**

STEP 2 **Open cover of BLT** to adjust gain settings.

STEP 3 **Select "Bal" input on the 250m².** Refer to the *Using the 250m² Balanced Line Input* section of the 250m²/500m Installation and Operation manual for detailed instructions.

STEP 4 **Turn gains of the source unit, BLT, and 250m² to their minimum positions. Disconnect speakers!!** Failure to disconnect speakers now will result in replacement of speakers later.

STEP 5 **Maximize output of source unit.** Using the Autosound 2000 Test CD #101 Track 99, set the source unit for the maximum undistorted output. If setting output levels without a measurement device such as an Archer Mini-Speaker or oscilloscope, turn source unit up to about 3/4 of the maximum level. Leave the source unit at this level for the rest of the adjustment.

STEP 6 **Play CD #101 Track 99.**

STEP 7 **Set Meter to read AC Volts.**

STEP 8 **Attach positive lead of AC voltmeter to test pad on circuit board.** *(Figure 11.2)*

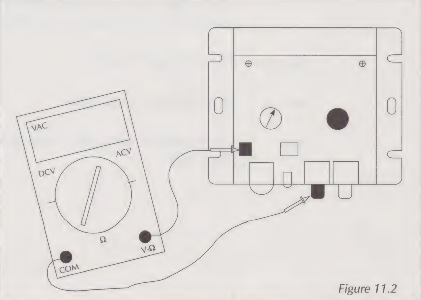


Figure 11.2

STEP 9 **Attach "Common" lead of AC voltmeter to RCA Shield of BLT.**

STEP 10 **Using the corresponding potentiometer, adjust the output voltage of the BLT to approximately 2.25 Vrms.** Perform the step for both channels of the BLT. ***Warning! Be sure that Track 99 time index reads greater than 30 seconds. If not, the adjustments will be incorrect.***

STEP 11 **Replace the covers of both the BLT and $250m^2$.** Refer to the *Using the $250m^2$ Balanced Line Input* section of the $250m^2$/500m Installation and Operation manual for detailed instructions.

- **ADJUSTING GAIN WITH BLT AND EPX_2**

STEP 1 **Install and wire source unit, BLT and EPX_2.** At this time, it is not necessary to wire and connect amplifiers. In fact, these adjustments can be made on a test bench before final installation into the vehicle.

STEP 2 **Open the cover of the BLT.**

STEP 3 **Turn gains of the source unit, BLT and EPX_2 to their minimum positions.** Amplifier connections are not needed at this time.

STEP 4 **Maximize output of source unit.** Using the Autosound 2000 Test CD #101 Track 99, set the source unit for the maximum undistorted output. If setting output levels without a measurement device such as an Archer Mini-Speaker or oscilloscope, turn source unit up to about 3/4 of the maximum level. Leave the source unit at this level for the rest of the adjustment. For detailed instructions on system level setting refer to the *System Level Setting* section in this chapter.

STEP 5 **Play CD #101 Track 99.**

STEP 6 **Set Meter to read AC Volts.**

STEP 7 **Attach positive lead of AC voltmeter to test pad on circuit board.** *(Figure 11.3)*

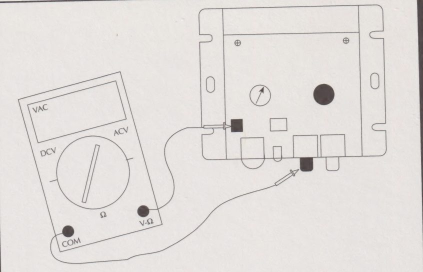


Figure 11.3

STEP 8 **Attach "Common" lead of AC voltmeter to RCA Shield of BLT.**

STEP 9 **Using the Corresponding potentiometer, adjust the output voltage of the BLT to approximately 2.25 Vrms.** Perform the step for both channels of the BLT. **Warning!** Be sure that Track 99 time index reads greater than 30 seconds. If not, the adjustments will be incorrect.

STEP 10 **Switch input jumpers from unbal. to bal. on the EPX_2 PC Board.** Refer to the *Optional Accessories* section of the EPX_2 Installation and Operation Manual for detailed instructions.

STEP 11 **Adjust EPX_2 input levels.** Set the input gains for both the left and right channels to 110 in the level setup menu of the EPX_2. Refer to the *Setup Level* section of the EPX_2 Installation and Operation Manual for detailed instructions.

STEP 12 **Replace the covers of both the BLT and the EPX_2.** The levels are now matched for the EPX_2, BLT, and source unit. Typically, amplifiers' input levels can be set to their minimum positions.

.T.T.I.®

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